TM

SOUND ON SOUND

Novation Supernova
Stellar Overdrive!

SUPERNOVA

Neuroglider
Prog A000

PLUS

Digital Data Reduction Explained

Emu Audity 2000

Pascal Gabriel

Guitar Recording Tips

& much, much more...

Creative Sequencing Tips

Roland SP808

Lexicon MPX100

WIN KORG TR-RACK SYNTH MODULE

ALLEN & HEATH GS3000 CONSOLE • ALESIS LX20 TYPE II ADAT
APOGEE AD8000 CONVERTER • AARDVARK AARK 20/20 PCI CARD
TC WORKS *NATIVE ESSENTIALS & EQ* • DBX DIGITAL DYNAMICS
JOHNSON MILLENNIUM GUITAR AMP • PC • APPLE • ATARI NEWS

9 770951 681061

Introducing the 2408.

The next breakthrough in hard disk recording.

The 2408 is a computer-based hard disk recording system that gives you 24 independent inputs and outputs • 8 analog inputs (with 20–bit 64x oversampling converters) • 8 analog outputs (with 20-bit 128x oversampling converters) • 24 channels of Alesis ADAT optical in and out • 24 channels of Tascam DA-88 in and out • 2 channels of S/PDIF in and out • 16-bit and 24-bit recording at 44.1 or 48 KHz • • digital dubbing between ADAT and DA-88 • digital dubbing between S/PDIF and ADAT or DA-88 • ADAT sync input • Word clock in and out • MOTU Control track input for sample-accurate Digital Timepiece synchronisation • includes the cross-platform PCI-324 card to connect the 2408 to your Mac or Windows computer • the PCI-324 will let you add additional 2408's for up to 72 input/output connections and as many channels as your computer allows • includes full-featured sample-accurate workstation software for Macintosh with recording, editing, mixing, and real-time 32-bit effects processing • • includes standard Windows audio (Wave) driver •

£995.00 INC VAT

19a HIGH STREET
SHEFFORD
BEDFORDSHIRE
SG17 5DD

TEL: 01462 812010
FAX: 01462 814010
Email: musictrack@compuserve.com

musictrack

Mark of the Unicorn, Inc   http://www.motu.com

leader

You may have noticed one or two guitar amp reviews cropping up in SOS in recent months, not to mention our regular reviews on guitar effects and processors, plus articles covering areas such as guitar recording techniques. Why? Partly because guitar amplifier technology has recently embraced physical modelling, which in theory allows one amplifier or preamp to emulate a number of classic guitar amp and speaker configurations in much the same way as a synth player accesses sound patches. However, it's also our view that the electric guitar remains a vitally important part of electronic music, not least because it's one of those instruments that can't successfully be imitated by a keyboard. Indeed, I can't even begin to imagine the complex web of real-time controllers that would be required to enable a keyboard to emulate all the subtle nuances of just a single picked guitar note.

Some manufacturers have tried to legitimise the guitar's role in the electronic music forum by building guitar synths, but although these devices have their place, all they really serve to achieve is to underline the fact that the majority of the guitar's subtleties are lost when the instrument is used to control synthesized or sampled sounds. In reality, you can get almost as wide a range of sounds by using effects to process the natural guitar sound, and this approach retains the immediacy and responsiveness of the instrument.

Of course, the future is undiscovered country and I know that a number of DSP engineers are pitting their wits against the problem of unravelling polyphonic audio performances to extract individual note information, so maybe the time will come when we have a guitar synth that has no tracking problems and where the synthesized sound takes on all the timbral nuances of the original. If this happens, it will probably be based on some type of physical modelling where aspects of the original string sound are mapped to specific parameters within the model. For example, when a guitarist 'digs into' a note to produce harmonics, this could be mapped to breath or overblow in a wind model. If and when all this comes to pass, I'll be the first in line to try it, and if all the usual guitarist's techniques of hammering, damping, stopping and squealing translate into some meaningful aspect of the sound being played, then I'll be well chuffed. However, I suspect that like the original guitar synths, the first wave of models featuring this as yet embryonic technology will fall far short of what the guitarist really wants, just as keyboard control over certain sound types is inherently limiting. The bottom line here is that as soon as you want to use a sound from a different instrument group to the one you're used to playing, you really need a new type of interface and new playing skills. For example, trying to play an emotional violin solo from a keyboard is as pointless as attempting a piano concerto using a MIDI violin.

There are numerous music composers who have worked entirely with electronic sound sources, but in the greater scheme of things, they have been largely unsuccessful. Despite what certain special interest groups may choose to believe, the vast majority of the civilised world remains unimpressed by Tangerine Dream or Kraftwerk, preferring instead music that combines electronic instruments with traditional instruments and voices. Fortunately, the addition of audio recording facilities to virtually all serious sequencing packages provides the means to experiment with a more balanced form of musical composition, but there's still the temptation to try to do everything from the keyboard. If you can resist that temptation and bring in a few 'real' players, you'll find that your music gains a lot more life and individuality. That even applies to dance music, after all, The Prodigy have a guitar player in the band and they haven't done too badly!

Paul White Editor

42 Lexicon MPX100

REVIEWS

Computer notes

in this issue

FEATURES

SOUND WORKSHOPS

REGULARS

90 Bon Harris

194 Tannoy Reveal Monitors

158 Pascal Gabriel: Peach

80 Roland SP808

70 Allen & Heath GS3000

218 dbx DDP

Korg TR Rack p266

PC MUSICIAN
Digital Transfer from Audio CD to PC   p104

Crosstalk

Musical May Day

In response to your July Sounding Off, I think Paul Nagle is right to be concerned at the general drift in popular music. I first noticed it during the '80s, when big-budget recordings started to sound the same. Big American rock bands, particularly, had a particular 'gloss' that tended to smooth out the differences (such as they were) between them. That process has accelerated and intensified to the point where a lot of popular music is identikit in nature.

Now, the relentless march of technology has pushed top-class recording within the reach of the average punter. It has never been easier to create halfway-decent music (and I emphasise the word 'halfway') music, and it will have enormous consequences.

The dystopian view looks like this: if you can't play, there are compositional aids that will generate music for you. And there has been an awful lot of progress in this area too. The other day I saw a soloist software tool demonstrated. It contains templates for the solo technique of a particular artist and then will generate a typical solo for a given set of chords. What would Stan Getz play over 'Take Five'? 'Ob-La-Di-Ob-La-Da'? 'Wonderwall'? Don't like Stan? How about Eric Clapton?

Let's extrapolate — give it 10 years and you'll load in your tune from your MIDI file collection, use the patches from the GM soundset and assign a soloist from the templates available. It's the logical end to a process that placed instruments, reproducible effects and compositional tools in the hands of the naive. It's the musical equivalent of Paint By Numbers, and it shows.

The utopian perspective is that, yes, all that will happen, but that motivated individuals (like me) will use the results to try out techniques and ideas before teaching them to a group of real musicians. Let's face it, being in a band is hard work — there are all those different, difficult people to work with. When it works, it's fabulous, better than almost any other feeling there is — when it works. When it doesn't it can be miserable. Anything to automatically shorten the process of developing a song is a good idea, but such advances will only ever furnish you with the foundations of an idea. The one thing I really liked about being in a band was that no matter how good the ideas I brought along

Trouble on the Cards

Can anyone help? I dabble in home recording on my PC; it's a very simple system, a 233MHz MMX with 64Mb of RAM, and I use an Creative Labs AWE64 Gold soundcard. The input from my guitar/keyboard goes into a Yamaha MU10 tone generator, and this is then connected to the Line In on my soundcard, the Line Out from the soundcard being connected to my hi-fi system. I use the MIDI output of the card with my cheapo Yamaha keyboard, and I have no problem recording MIDI. The problem comes when I try to record guitar or keyboards. 90 percent of the time, I get a horrible feedback-type noise, and even before I start playing anything the level meters are way in the red. The weird thing is that recording does work the other 10 percent of the time, although it's then always very quiet (I get around this, though, by editing the recording in *Sound Forge*). I think I've tried everything with the levels I can change, so any further suggestions would be very welcome. The software I have been using is *Cubase VST* and *Cakewalk Pro* along with Sonic Foundry's *Acid*. The results are exactly the same in each case.
Brian Rooney
via email

Editor Paul White replies:
I haven't heard of this specific problem before, but I suspect that you have in some way set up a feedback path so that some of your audio input is being fed back to itself. If so, it's unlikely that your choice of software will have any effect on this problem. The fact that your

The Yamaha MU10 only has a line level input, so it's not ideal for direct recording of electric guitar.

meters are already registering before you start to record indicates that something of this sort is wrong, though without an exact diagram of your setup, I can't be sure. However, even if you find a way around the feedback, you'll also suffer a level and impedance mismatch when trying to record guitar in this way, as the MU10 inputs aren't particularly sophisticated. This would result in low levels and unnecessary noise, so at the risk of suggesting you spend more

money, I really think you should consider buying a guitar recording preamp or effects unit and then patch that directly into the audio input of your soundcard. I'm also a little unsure as to why you're recording the output from your keyboard rather than running it live from MIDI.

In your position, I'd feed audio directly into the soundcard (via a suitable preamp where necessary), then feed the soundcard output to the stereo line input of the MU10, enabling you to mix the audio recordings with your sequenced MIDI parts. Playing all the MIDI parts back from the sequencer at mixdown rather than recording them should produce better sound quality as well as saving a lot of hard disk space. ❏

were, they were always improved by the others working on it.

I think this is why guitar bands remain prominent; the guitar is (relatively) easy to learn, and it still carries the player's trademark sound, even though programmable effects units have had their influence here too. Want to sound like

Brian May? Get a Brian May-endorsed guitar, a good clean-sounding combo and an effects unit that has Brian May's settings programmed into it. See? Even the guitar is being slowly assimilated. Resistance is futile!
Richard Lamont Abrams
via email ❏

Crosstalk

DAT's the way to do it

I have a couple of comments concerning the June issue of SOS. In that month's Crosstalk, Steve Bradbury asked how he could back up PC data to his audio DAT machine. I appreciate that Steve doesn't really want to buy yet another audio card, but other readers may be interested to know that the Zefiro Acoustics ZA2 digital I/O card comes complete with a DOS utility to back up PC data via S/PDIF or AES/EBU to any audio DAT machine. The software includes extra error correction, and will work fine with audio DAT tapes. However, I would not recommend buying a ZA2 just to back up PC data — a CD burner would make better economic sense for this.

In his reply, Martin Walker talked about the difference between 'audio' and 'data' DAT tape. This is in danger of becoming one of those urban myths — the only differences between the two are the shell printing, the packaging and the price. The cases of both types have identical holes for selecting tension, and we have successfully used both types for a number of years in our audio DAT machines with no ill effects. It's probable that the data type undergo more rigorous inspection to guarantee quality, and if so, this would be a bonus for audio use — however, in our experience, failure and/or error rates are similar for both types, which at

least implies that the actual tapes come straight off the same production lines!
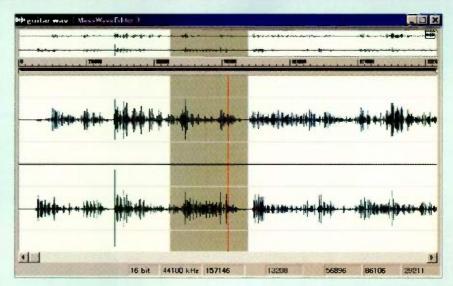
Data DATs are marked by length; a 60m tape is the equivalent of just over 2 hours, and a 90m tape over 3 hours. 120m data DATs are also available (4+ hours) but these use thinner tape, with which many audio DAT machines are not designed to cope. So, apart from avoiding the 120m data DATs, my advice is buy whichever you can get at a good price. For example, I just picked up a couple of TDK 90m tapes on offer at PC World for £2.24 each including VAT!
Rob Kirkwood (proprietor, RKMS, Nottingham) via email ❑

Time for a Change

I have a plan to slightly change my studio setup, which is at present based around a Mackie 1604 and a Fostex DMT8VI, all sync'ed through a PC running Cakewalk. My present sampling requirement is satisfied by a Tropez card on the PC, but I would like to change to a stand-alone sampler and would like this to have eight outputs for processing through the Mackie. However at the moment all 16 inputs are filled (eight with various bits and bobs and eight with the direct outs from the DMT8VI).

I have an idea how I can get round this without using aux returns, or indeed by buying a bigger mixer, but would welcome your thoughts as to how successful this might be. My first thought is to do things the other way round; ie. use the DMT8VI as my 'final' mixer by plugging the Mackie main stereo outs to aux ins on the DMT8VI. Thus all MIDI equipment would be submixed in the Mackie then added live to the digitally recorded stuff for final mix within the DMT8VI. This would free up eight inputs on the Mackie by taking the direct outs from the DMT8VI.

I think this will work: my question is, do you think it will increase the noise level to the final mix? Can you see any glaring problems? I guess one might be how to work out the best route for external processing. I use a Lexicon Alex as main reverb (which could obviously be patched in on the DMT8VI so as to process everything, but at the cost of flexibility of effecting different channels on the Mackie), a Zoom for other things, and a Behringer Ultrafex to generally sweeten everything. Any suggestions as to where to place these within the proposed setup would be very welcome. The stuff I record on to the DMT8VI is mainly acoustic (guitar and vocals) and is normally recorded flat with effects added at final mix. My other thought was to keep all effects units within the Mackie (as this is where I need to

Akai's Mesa II — available for the PC at last.

use it most) and buy another effects unit as a final additional processing for the complete mix.

My only other question regards sampling. If I swtich to an Akai stand-alone sampler, I will miss the visual editing associated with my Tropez card. How does Akai's Mesa sample editing software compare, and does the sample have to be loaded into the computer for editing, or is it done remotely (my PC does not possess SCSI)?

Thanks for a great magazine as ever.
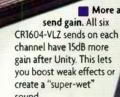Craig Alexander
London

Editor Paul White replies: *We've all run out of mixer inputs at one point or another, and using a submixer as you suggest is actually quite a good way to deal with the problem. Though the noise figure will never be better than that of the final mixer in the chain, you shouldn't have too much of*
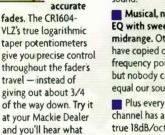
a problem as everything is being mixed at line level. However, figuring out how to deploy effects can be a bit of a challenge. One solution to this is to use an effects unit that has a stereo input but which mixes both inputs to mono to feed the effects. In fact, most effects units work this way, unless they're set to dual-effects mode, and your Alex should be fine. The trick is to set the Alex to 100% wet and feed one input from your submixer's aux send and the other from your multitracker's aux send. Take the Alex outputs to your Mackie's aux returns and feed the Mackie into the aux returns on your multitracker as you suggest.

On the subject of Akai's Mesa software, you will need SCSI to get the samples into your PC, but it is worthwhile, as it enables you to use inexpensive WAV file sample CDs and then convert them into pukka Akai-format programs. I've only used Mesa for the Mac, but apparently the official PC version is finally available; contact Akai UK for details (0181 897 6388). ❑

shape of things to come

by Derek Johnson

Yen for Cheap Samplers

Sampler and digital recorder specialists Akai chose the recent Docklands-based Mad About Music show as the venue for one very interesting announcement. Perhaps related to a recent drop in the value of the Yen, and possibly also to a restructuring of the sampler range due to

£650, and that the S3000XL has had another £400 sliced off its price. Akai's new samplers will also actually cost less than first thought. Those new prices are as follows:

- S6000 (due October 1998): £2799
- S5000 (due September 1998): £1799
- S3000XL: £1099
- MPC2000: £899
- S2000 Studio (with 8-output and digital I/O board): £849
- S2000: £649
- Remix 16: £549
- S20: £399

A range of accessories for the new S5000 and S6000 has also been finalised. Standard equipment on the S6000, but available as options for the S5000, are the EB20 20-bit multi-effects board (£299), IB509P 8-output expansion board (£249), and the VOX64 64-voice expansion board (£349). The IB1616A 16-channel ADAT-format digital I/O board (£299) is an option for both samplers, while an internal 1Gb Jaz removable drive (£299) is available for the S6000 only.

the imminent launch of two new samplers, Akai have implemented a series of price cuts. The bottom line is that you can now buy an S2000 — which recently listed for £799, down from £899, after last October's price cuts — for less than

A Akai UK, Haslemere Heathrow Estate, The Parkway, Hounslow, Middlesex TW4 6NQ, UK.
T +44 (0)181 897 6388.
F +44 (0)181 759 8268.

G Tek hit the Spot

Repair specialists G Tek have a soft spot for valve gear; valve amplifiers in particular are a speciality. Any seasoned user of valve equipment knows that valves eventually wear out, and as they age the sound of your amp is affected. Rather than simply replacing valves one at a time, when they finally blow, G Tek recommend revalving and rebiasing, which involves a regular renewal of all valves so that they're perfectly matched and working at full

efficiency. G Tek select and electrically match all valves, and set amps up once the valves are installed.

In addition to this service, G Tek can repair, modify, hot-rod or renovate amps of any age, repair or customise effects pedals, and re-cone or repair loudspeakers.

A G Tek, 10 Barley Mow Passage, Chiswick, London W4 4PH, UK.
T +44 (0)181 994 6477 ext 2133.
F +44 (0)181 987 9077.
E GTek@jgtek.demon.co.uk

Jarre Jam

Sunday, August 23 is the date for this year's JarreCon, the fourth UK Jean-Michel Jarre convention, taking place at the Guildhall in Birmingham's Aston University. Highlights of the day will include exhibitors selling merchandise, CDs, records, T-shirts, posters and photos, a video screen playing rare interviews, videos and concert footage, a quiz to find the 'Jarre Fan of 1998', and an auction of rare records and merchandise, with proceeds going to UNESCO (of which JMJ is a Goodwill Ambassador). Representatives of Jarre fanclubs from all over Europe will be at the convention, and a DJ will be taking requests throughout the day.
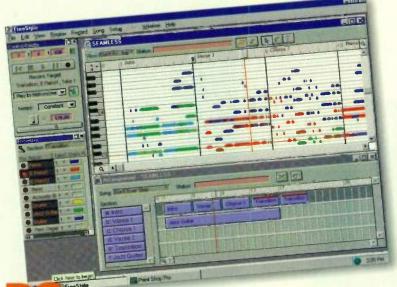
The most unusual happening of the day will probably be the theremin demonstration. A unit will be available for visitors to listen to and play, with a backing track of *Oxygene 10* provided for fans to jam to; an approximate time of five minutes will be allocated to each person, for the fee of £1 — which will, again, be donated to UNESCO.

Advance tickets are available at £7.50 each, £14 for two, £20 for three, and £6 each for bookings of four or more. In addition, programmes and JarreCon T-shirts can now be pre-ordered along with tickets. And everyone who purchases an advance ticket will be entered into a draw for exclusive prizes.

A Natural Image Productions, 76 Etwall Road, Hall Green, Birmingham B28 OLE, UK.
T +44 (0)121 693 2477.
F +44 (0)121 693 2477.
E jarrecon@jarre.force9.co.uk
W www.jarre.force9.co.uk

PC breaks free

Version 2.0 of *Freestyle*, Mark of the Unicorn's trackless sequencer with instant notation, is now available for PC (v2.01 for Mac was reviewed in the May 1998 issue of *SOS*). *Freestyle* aims to get in the way of composition much less often than most sequencers, and rather than working with tracks, program changes and so on, provides you with on-screen Ensembles, Players, Takes and Arrangements. The

tempo at which you play is sensed automatically, no matter how *rubato* you play, you have complete freedom over where to place time signatures, and your music shows up on screen as you play, pretty much exactly as it will print out. The software comes with built-in support for dozens of popular MIDI instruments, including any General MIDI device, and requires a 486/66 MHz or Pentium-based PC (or faster) with at least 16Mb of RAM, running Windows 3.1 or Windows 95. A Pentium PC with at least 24Mb of RAM is recommended.

A MusicTrack, 19a High Street, Shefford,
  Bedfordshire SG17 5DD, UK.
T +44 (0)1462 812010.
F +44 (0)1462 814010.
W www.motu.com

Pro Tools goes Knobular

ProControl is a modular, expandable hardware control surface that adds tactile mixing and editing capability to Digidesign's Pro Tools TDM systems, especially the flagship Pro Tools 24 system. It interacts with Pro Tools software via DigiFader moving faders, 25 high-resolution, 8-character "scribble strips" and dedicated switch and encoder controls — including control-room mix monitoring and talkback. ProControl connects to Pro Tools TDM systems via a low-cost, standard Ethernet connection.

The base ProControl unit provides everything you'll need to run any Pro Tools session, small or large. The surface has three basic sections — fader section, main section and integrated meter bridge — and while the largest Pro Tools TDM mix configuration may be accessed by the base unit, there are applications where you might want additional Fader Sections, so ProControl can be expanded in 8-channel increments to a maximum of 32 channels. Each expansion unit contains all the features of the Fader Section and also includes eight stereo meters.

A Digidesign UK, Avid Technology Ltd, Westside
  Complex, Pinewood Studios, Iver Heath, Pinewood,
  Bucks SLO 0NH, UK.
T +44 (0)1753 653322.
F +44 (0)1753 654999.
W www.digidesign.com

Logic Audio take note

Almost before the software is available, users of Emagic's new *Logic Audio Gold* and *Platinum* MIDI + Audio sequencing software can take advantage of a tutorial book. Johannes Prischl (he's behind the notation tutorial on the *Logic Audio* v3 CD-ROM) has self-published the *Logic Notation Guide*. The book runs to 204 pages, and comes with a floppy disk (Mac or Windows 95 format) containing 28 *Logic* song files. It covers musical notation with *Logic Audio Gold* and *Platinum* software on a medium to advanced level, using an approach based on practical examples, many of which are included as *Logic* files on the disc. Apart from presenting different approaches for working with *Logic*'s

Score Editor, both for MIDI sequencing and score printout, the book also contains some special solutions for notation effects which seem to be impossible using the basic functions described in the manual. The *Guide* covers both the Mac and Windows versions of the software, and differences between the versions are pointed out

The *Logic Notation Guide* costs 495 Austrian Schillings; that's about £24. Postage is extra, of course, and a variety of options are available. The non-priority European Union rate is 66 schillings (about £3.30), while European priority is 110 schillings (£5.40). Austrian readers pay just 49.50 schillings, while the rest of the world (non-EU) pays 60 schillings non-priority; priority shipping is

available for non-EU countries at 100 schillings, and for the rest of the world for 150 schillings. Payment can be made by credit card or Eurocheque, but contact Johannes first for further details.

Finally, just as we were going to press (early July), we received news from UK Emagic distributors Sound Technology that the new 'metallic' versions of *Logic* are now shipping after a short delay. *Logic Audio Gold* and *Logic Audio Platinum* are the first out of the starting gate, but *Logic Audio Silver* is still to come. More on these next

month, and as usual, look out for *SOS* reviews shortly!

A Johannes Prischl,
  Schoenburgstrasse 19/17,
  A-1040 Vienna, Austria.
F +43 1 504 2771.
E jprischl@t0.or.at
W www.t0.or.at/~jprischl/LNG_E.htm

A Sound Technology,
  Letchworth Point, Letchworth,
  Hertfordshire SG6 1ND, UK.
T +44 (0)1462 480000.
F +44 (0)1462 480800.
E info@soundtech.co.uk
W www.soundtech.co.uk

Some Pictures From The Book....

(go to top of page)

In the book there are detailed explanations how these scores can be created. In addition the scores shown here are also included as Logic files on the supplementary disc.

(Note: If you don't see the pictures, click on the Reload button in your browser.)

Beamed Grace Notes

shape of things to come

DACS ring the changes

Following the successful launch of their Clarity range of high-quality signal processors, DACS Ltd have made an idiosyncratic move by releasing a family of high-quality rackmounting ring modulators. Simply put, a ring modulator produces an output based on the sum and difference frequencies of two audio inputs — typically, a pair of waveforms or a music source and a modulating waveform.

DACS' FwS series comes in five flavours:

• The MODual8 ring modulator (£299 including VAT) offers two inputs and one output, with spectral controls for 'Weight' and 'Edge'.

• The MODual8-2 (£399) offers *two* ring modulators, with auto-switching to allow either 'modulator' input to operate on both 'music' inputs.

• The ColOSCii (£369) is a single ring modulator with built-in oscillator, five frequency bands, and Tune and Fine Tune controls.

• The ColOSCii-2 (£469) offers two of the above, with the oscillator switchable to feed both modulators.

• Lastly, the FREQue (£599) is two ring modulators with two built-in oscillators, a CV input for both oscillators and a frequency shifter.

A DACS Ltd, Stonehills, Shields Road, Pelaw, Gateshead NE10 0HW, UK.
T +44 (0)191 438 5585.
F +44 (0)191 438 6967.
E info@dacs-audio.co.uk
W www.dacs-audio.co.uk

Ensoniq's Capital Gains

Ensoniq have released a couple of enhancements to the Paris digital recording system (reviewed in *SOS* January 1998). The most significant aspect of the new (and free) v1.56 software update is the addition of support for the new ADI-01 ADAT digital interface module for Paris' modular expansion chassis (MEC). The new module features two optical connectors that allow the digital transfer of eight tracks of audio to and from a Paris system; a 9-pin sync interface allows the synchronisation of up to 16 ADAT recorders directly from Paris. The new software can be downloaded from the Ensoniq web site.

A Key Audio Systems Ltd, Unit D, 37 Robjohns Road, Chelmsford, Essex CM1 3AG, UK.
T +44 (0)1245 344001.
F +44 (0)1245 344002.
E info@keyaudio.co.uk
W www.paris.ensoniq.com

SOS can be reached at
sos.feedback@sospubs.co.uk

Our web site address is
www.sospubs.co.uk

SPIRIT by Soundcraft

shape of things to come

Netscape: Some more Pictures

Location http://www.cameo.nl/Fenix/smallptx.html

Fénix being tested

Modular Synth goes Dutch

When it comes to classic analogue synths, it's probably true to say that the majority are of American or Japanese manufacture. Yet Dutch company Synton produced a couple of classics during their lifetime — in fact, the Synton Syrinx monosynth rarely comes up for sale, and is almost as desirable, and pricey, as a Minimoog. It's interesting to note, therefore, that the people behind Synton are actually hand-building, in their spare time, a modular synth directly descended from Synton's 3000 modular synth and the Syrinx. The Fénix, as the synth is called, has been designed as a compact, rackmounting instrument with a fixed set of separate modules that need to be patched together to produce a sound — rather like a Korg MS50 or Roland 700. There are 65 rotary controls on the modules across the top of the front panel, with about 160 banana sockets on the lower half. The unit is about five inches deep and 9U high. One other nifty feature is that almost all the modules are deliberately different from each other: every LFO and Envelope Generator has different controls and connections, as do most of the mixers, VCFs, and so on.

The Fénix includes three VCOs, three VCFs, three LFOs, three EGs, four VCAs, various mixers, a slew limiter, sample & hold, portamento, noise generator, two ring modulators, CV mixers and two audio-in processors. The stereo audio out module can also drive a pair of headphones. User reports on the Fénix web site seem to indicate that the synth has some unique and unusual patching possibilities, and since there are none of the now-scarce Curtis chips in the unit (as used by the Syrinx), parts are readily available, for ease of repair and servicing in the future:

If you're interested, you'll be pleased at the surprisingly low price: instruments are sold direct (because, as a spokesperson says, "we'd like to know where these machines end up"), which keeps prices down to a fairly reasonable 3200 Dutch Guilders (less than £1000 or US$1600). Manuals and 230V power supply are included in the price, but delivery and patch cables are not: a 12U flightcase is available as an option. However, since every Fénix is hand-built by a team with other commitments, they're never going to be plentiful (the first run was limited to 25, but the team are planning to build a few more) and you'll have to be prepared to wait for your synth.

A Marc Paping, Willem de Zwijgerlaan 256, 1055 RE Amsterdam, Holland.
E marc.paping@cameo.nl
W www.cameo.nl/Fenix/index.html

Furman: Something Old, Something New...

A new release from Furman Sound is also a blast from the past: the PQ3 single-channel parametric EQ/instrument preamp was actually Furman's first product, launched in 1976. It was one of the first rackmount products developed specifically for musicians, and has apparently developed a cult following. The new version is identical to the original, save for the addition of a front-panel input. It offers three bands of peaking EQ (using constant-Q EQ curves), each with controls for centre frequency, bandwidth and boost/cut. The three frequency bands overlap, giving the PQ3 a total range of 20Hz-20kHz. There's also an EQ bypass option and EQ In, overload and power indicators, and connections are on high- or low-level jack inputs, though the PQ3B

balanced processor, with both XLRs and jacks, is on offer if you prefer balanced connections.

Furman have also introduced two new patchbays: the PB48 offers 48 TRS jacks, front and back, that may be used for balanced or unbalanced systems. The PB48D offers D-Sub connectors on the rear panel. Half-normalling is standard, and may be defeated without soldering or track cutting.

Last up, we have news of Furman's new headphone distribution system. The HDS6 is a rackmount system which connects to your mixing console and drives a chain of HR6 personal headphone mixers. Each HR6 provides a musician with five volume controls — four mono (for mixer channels or busses) and one stereo, for

main or effects mix. Linking between HDS6 and the HR6s is via affordable Ethernet connectors and cabling.

A Audio Projects, Unit 8, Speedgate Farm, Mussenden Lane, Fawkham, Kent DA3 8NJ, UK.
T +44 (0)1474 879446.
F +44 (0)1474 872925.
E sales@audiopro.force9.co.uk
T Audio Projects +44 (0)1474 879446.

1998 CATALOGUE

World Radio History

DDrum find the Key

Key Audio Systems have announced that they'll be distributing the ddrum4 electronic drum system from Swedish manufacturer Clavia — whose name you may know better from the acclaimed Nord Lead family of synths. The company has actually been involved in electronic percussion since 1983, with the ddrum name well-established before the launch of the Nords.

The new ddrum4 includes an advanced percussion 'brain', plus "real-feel" kick, tom and snare controllers. In addition, Clavia have pioneered cymbal and hi-hat controllers, with rubberised metal plates, which can be played just like the real thing. Ddrum4 offers a large memory capacity and compact design, and Clavia claim good triggering response and fine sound reproduction.

Returning to the Nord family for a moment, we've just heard that the Nord Modular synth (reviewed in *SOS* April and May 1998) has had a software upgrade. Version 2.0 is now available free of charge from the Clavia web site, and offers new modules and improved editing functions. One noteable new feature is the 16-band vocoder, the result of "months of research with some famous analogue predecessors". Clavia claim that the vocoder sounds and behaves just like the best analogue models. Another new feature of v2.0 is the complete drum synthesizer module, which provides the user has with pre-programmed templates for typical analogue drum sounds. New audio modifiers include a phaser with positive and negative feedback, a compressor/limiter and an expander/gate, and new filters include parametric EQ, bass, and treble, while a formant-wave oscillator joins the oscillator modules.

The v2.0 upgrade offers new tools, too, such as copy/paste and undo.

A Key Audio Systems Ltd, Unit D, 37 Robjohns Road, Chelmsford, Essex CM1 3AG ,UK.
T +44 (0)1245 344001.
F +44 (0)1245 344002.
E info@keyaudio.co.uk
W www.clavia.se

Production team Clive Langer and Alan Winstanley — featured in *SOS* last month — have taken delivery of another pair of Spirit Absolute Zero nearfield monitors. Clive actually has two pairs at home: "They have a real fullness of sound and lots of body. What can I say? They're really good!" Alan notes: "I always prefer to mix on monitors that sound like the speakers listeners have at home. With the Absolute Zeros that's exactly what you get."
T +44 (0)1707 665000.

An error crept into last month's Shape Of Things To Come; on page 16, SCV's contact details were given in connection with an item about new TC Works plug-ins for the Macintosh ('Mac Gets The Works'). The correct company to contact for further information regarding TC Works is of course Arbiter, whose details are given below. Our apologies to both companies.
A Arbiter Music Technology, Wilberforce Road, London, NW9 6AX, UK.
T +44 (0)181 202 1199.
F +44 (0)181 202 7076.
E arbiter@cix.compulink.co.uk
W www.arbitergroup.com

The IR1 Mic Mute, from British company Missing Link Music, is a lightweight, proximity-sensitive microphone switch that is activated only when a performer moves into its range. The mic remains switched off when no-one is near it, minimising both unwanted backline pickup and the possibility of feedback (if feedback does occur, the performer can put a stop to it simply by moving out of sensor range). The IR1 is powered by a 6V battery which will apparently last for 72 hours in continuous use.
T +44 (0)1293 887512.

Amateur DJ Phil Whitehead, from Poynton on the outskirts of Manchester, was the lucky winner of a Roland MC505 Groovebox at the recent DJ Culture show in Manchester. His entry was drawn during a Roland Remixing workshop.
T Roland Brochure line +44 (0)1792 515020.

DAT's a First!

The new DA45HR is not only a first for Tascam: it's the first high-resolution 24-bit DAT recorder to come to the marketplace at all. The 3U rackmounting machine records true 24-bit digital audio onto an ordinary DAT tape, with one catch: recording time is halved, so that a 120-minute tape will provide 60 minutes of 24-bit audio. Should you need the facility, standard 16-bit recordings can be made, and 16-bit tapes can be played back on the DA45HR.

The DA45HR features 24-bit analogue-to-digital converters and 20-bit digital-to-analogue converters with dither, and will integrate into virtually any audio production environment, analogue or digital. The balanced XLR and unbalanced phono analogue inputs and outputs are joined by both AES/EBU and S/PDIF digital I/O, enabling the recorder to communicate with digital mixers and recorders; Word Sync In and Thru allow for easy connection to digital recording consoles, and there's even a 15-pin parallel interface for added control via fader starts or GPI (General Purpose Interface). The user interface, including subcode handling, should be familiar to all DAT machine users, though a new menu-driven software architecture features an easy-to-read display that aims to promote "intuitive operation". The recorder's menu layout makes it easy to access and edit the numerous system parameters, and the machine even provides alpha-numeric character entry for naming program selections. To aid searching through program material, a shuttle wheel and a 2-position memory/locate function are provided. The RC-D45 Remote is available as an option.

A Tascam UK, 5 Marlin House, The Croxley Centre, Watford, Hertfordshire WD1 5YA, UK.
T +44 (0)1923 819630.
F +44 (0)1923 236290.
W www.tascam.com

The list of new features is endless...

- Resolution 15360 ticks ppqn
- User definable key commands
- Screensets
- Dedicated controller editor
- Audio automation editor
- Enlarged toolbox
- 4 Arpeggiators
- Groovebox quantize module
- VST style Midimixer
- Markers
- Folder tracks
- Polyphony reduction
- Drag & drop from desktop
- Customizable control bar
- Up to 64 audio tracks
- Hi quality EQ
- Insert points on every channel
- 8 aux effect slots
- Audio sub groups
- Unlimited Logical editor presets
- Long track & file names
- Extended Inspector
- Randomise midi data
- Preset select by name GM/GS/XG
- Extensive preference options
- 4 choices of transport bars
- Groove quantize select tool
- Advanced track control
- Near sampleaccurate non-destructive audio editing
- DSP processing available from arrangement page

So why not get ahead of the game and book a course with one of the approved training centres.

The approved Steinberg training centres have all passed the demanding criteria set by Steinberg in Germany. The facilities for teaching are excellent and the quality of course material is very high. Courses generally start in September so book now to secure a space.

The Academy Of Contemporary Music	Guildford	01483 456 888	
Jewel & Esk Valley College	Edinburgh	0131 657 7235	gwcir@jevc.ac.uk
Manchester MIDI School	Manchester	0161 953 4072	mms@mmr1.poptel.org.uk
Metro New Media	London	0171 729 9992	info@mnm.co.uk
Newcastle College - Digital Solutions	Newcastle	0191 200 4727	sss.ncl-col.ac.uk
SAE Technology College	Glasgow	0141 221 3441	enquiries@sael.demon.co.uk

The 6 centres listed here all teach on the Mac platform. Please call us for a list of PC centres.

Distributed exclusively by ARBITER GROUP PLC Wilberforce Rd. London NW9 6AX
Tel: 0181 970 1909 • Fax: 0181 202 7076 • Web: www.steinberg.net

Steinberg

CUBASE
MUSIC RECORDING AND EDITING SYSTEM
4.0

shape of things to come

Share & Enjoy

A number of applications have been written for, or support, *MIDIShare*, a nifty cross-platform, real-time, multitasking, operating system developed by French research group Grame.

Mil Products' *Modularing* is the latest, and is basically a modular suite of interactive analogue-style tools that allows you to treat MIDI notes and data as if it were living in an electronic music studio of the late '60s. Don't be fooled by the on-screen graphics, which may lead you to presume you're playing with a DSP-based virtual synth of some kind: you're always dealing with MIDI data, not digital audio. However, it is possible to have *Modularing* play a QuickTime-based synth. First of all, you need

MIDIShare v1.68, but this comes with the complete *Modularing* package; if you have *MIDIShare* already, download the smaller version of *Modularing*.

The tools provided by *Modularing* (which runs on Power Macs only) can be freely interconnected, allowing you to build complex applications using quite simple building blocks. The collection currently includes:

- Sequencers: Seq Analog X8; Seq Matrix X16; Seq Pad X4
- Triggers: Trigger Mono X1; Trigger Poly X4
- Effects: Arpeggiator; Delay; Delay Note Shifter X4
- Controllers: Bulker, Keyboard, Mix Controller; Tempo Controller
- Quicktime Synth Expander

Coming in the future will be comprehensive MIDI remote control options, whereby each module has a full MIDI spec,

allowing MIDI controllers to be used to change parameters. Not only will applications on a single computer be able to control each other using MIDI, but applications networked on several connected computers will also be able to interact.
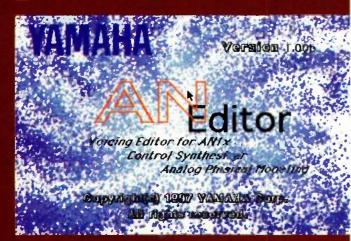
And the best news is that the whole thing is free!

AN1x gets Mac attack

The latest piece of free software from Yamaha is a full AN1x editor for Macintosh. The Mac software makes the AN1x (previewed and reviewed in *SOS* April and August 1997) even more accessible, and has a comprehensive user interface that allows the synth settings to be controlled and stored with ease. And if you need help, there's an on-line manual that provides answers at the click of a mouse. The software can be downloaded from Yamaha's web site.

Carry on Mixing

If you're looking for the all-in-one mixing solution for onstage use, Yamaha believe they have the answer in their new EMX2000 powered mixer, which features an 'Energy Efficient Engine' power amp. Two channels, yielding 200W apiece, are on board, together with digital effects and a 12-input mixer with XLR mic and jack inputs, peak indicator, gain control, 3-band EQ, two monitor sends, one effects send, PFL and a level fader. Inputs 1-4 have insert points, and there's a 7-band graphic EQ on the stereo output. Two stereo sub inputs allow various sound sources to be integrated, and a 'Rec' out allows you to easily record your performances to a stereo machine. The whole package weighs just 14kg (that's 30lbs in old money) and has a useful carrying handle.

And what a coincidence: Yamaha have also launched a new range of PA speakers, designed in the USA. The American Series consists of a wide

variety of cabinets, at entry-level and mid-range price points. All cabinets are trapezoidal in design, with inputs via two standard jacks (making it easy to link multiple cabinets), and all speakers are rated at 8Ω. The range can be stacked, or mounted on stands, courtesy of in-built 'top hat' connectors on the bottom of the cabs.

If you're in need of something to put that PA on, Yamaha can also help: they're now handling the UK sales, marketing and distribution of the Ultimate Support range of instrument stands. The range includes the Apex range of keyboard stands, plus PA speaker tripods, mic stands, monitor stands, and more.

The first name in monitoring Headphones...

More hit records and sound tracks have been mixed and monitored on beyerdynamic's range of headphones than any other headphones in the world.

Our range of headphones are acknowledged by recording engineers and hit making musicians for their accuracy and fidelity. ■

The first name in monitoring headphones — beyerdynamic... *Fidelity in Audio.*™

for more information
Tel. 01444 258258
e-mail: sales@beyerdynamic.co.uk
Burgess Hill, W.Sussex RH15 9TN

beyerdynamic))))

shape of things to come

Sweet M.A.

I don't suppose many of you have been waiting anxiously for news of Master's degrees in music technology-related subjects, but rather like London buses, details of several have turned up at the same time this month. All three MAs start this September, and all are available as one-year full-time or two year part-time courses.

First up is the University of Westminster's Audio Production MA program, which promises "a new type of education for audio specialists". The MA aims to develop creative abilities in audio, across music, sound, radio, TV, film and multimedia, all with a firm basis in the technology involved. Applicants will need to have a high level of experience in audio production and demonstrate this through a portfolio of audio production work.

Then there's Goldsmiths College's MA in Contemporary Music Studies. Available in a one-year full-time or two-year part-time form, the programme combines the study of musical repertoirs and techniques with the cultural theory behind them, and includes the study of popular as well as contenporary classical music. The programme joins five existing MAs offered by Goldsmiths' music department. Note that the department's facilities include an electronic music studio, and that several members of staff have expertise in "new technologies in music."

Last of all, Kingston University's School of Music is offering a new master's course in Composing for Film and TV (subject to validation). It will be taught by Richard Harvey (his credits include GBH, Jane Eyre, and The Ambassador) and Paul Pritchard (Dr Finlay). There will also be input from visiting professionals, such as Michael Nyman and Patrick Doyle (Donnie Brasco). The course will focus on techniques of putting music to picture, and will make use of Kingston's studios and instrumental ensembles. Opportunities will be available for working with the design faculty's film-makers and animators. Course modules include Composing for Film and TV, Studio Composition, and History of Film Music.

A load of Koblo?

The first release from new Danish software company Koblo is the *Vibra9000* monophonic synth for Power Macs. The company's goal was not to emulate old analogue instruments, but to produce "an entirely new, powerful digital instrument", with no extra hardware required. *Vibra9000* offers a wide range of new oscillators, filters, envelopes, LFOs and modulators, plus arpeggiator, hard disk recording, true stereo operation, full MIDI control, and more. Much care has gone into the user interface, providing easy access to all parameters. A Windows version is expected later this year. Free trial versions and detailed information can be found at the Koblo web site.

- **A** Koblo, Hvesager 47, 7300 Jelling, Denmark.
- **T** +45 7680 1040.
- **F** +45 7680 1041.
- **E** max@koblo.com.
- **W** www.koblo.com

- **A** University of Westminster, Watford Road, Harrow, Middlesex HA1 3TP, UK.
- **T** +44 (0)171 911 5903.

- **A** Goldsmiths College, New Cross, London SE14 6NW, UK.
- **T** +44 (0)171 919 7660.
- **F** +44 (0)171 919 7644.
- **E** s.mcveigh@gold.ac.uk

- **A** Mike Searby, School of Music, Kingston University, Kingston Hill, Kingston-upon-Thames, Surrey KT2 7LB, UK.
- **T** +44 (0)181 547 7149.
- **F** +44 (0)181 547 7149.
- **E** m.searby@kingston.ac.uk
- **W** www.kingston.ac.uk

If you're fed up with wall-warts trailing around your studio, check out MTR's DC8DPR, a fully regulated 8-way 9V DC power supply for guitar pedals, radio mic transmitters and certain studio outboard equipment. The polarity of each output can be changed, and a twin-colour LED above each pair of outputs indicates the chosen polarity. The unit has a current capability of 4.1 Amps, and costs £89 including VAT.
T +44 (0)1923 234050.

John Hornby Skewes have introduced a new range of cables, under the banner 'Connected'. The range includes 6-foot, 10-foot and 25-foot 8-way phono-to-jack looms; 3-foot, 6-foot, 10-foot and 16-foot jack-to-jack looms; balanced mic cables; patch-cable packs; and MIDI leads ranging from one foot to 20 feet in length. Gigging musicians can choose from 12-, 20- and 24-way, 100-foot, XLR-equipped stage boxes. Speakon connector-equipped speaker cables, and more.
T +44 (0)113 286 5381.

A new version of the Joemeek SC2 compressor is due, so Joemeek Distribution are reducing the price of the current SC2 to £1056; this reduction will remain in force after the introduction of the new model in September. The new SC2 will add a 'Slope' switch position, and a VU meter switch; its connectors will be XLR only.
T +44 (0)1626 333948.
E sales@joemeek-uk.com
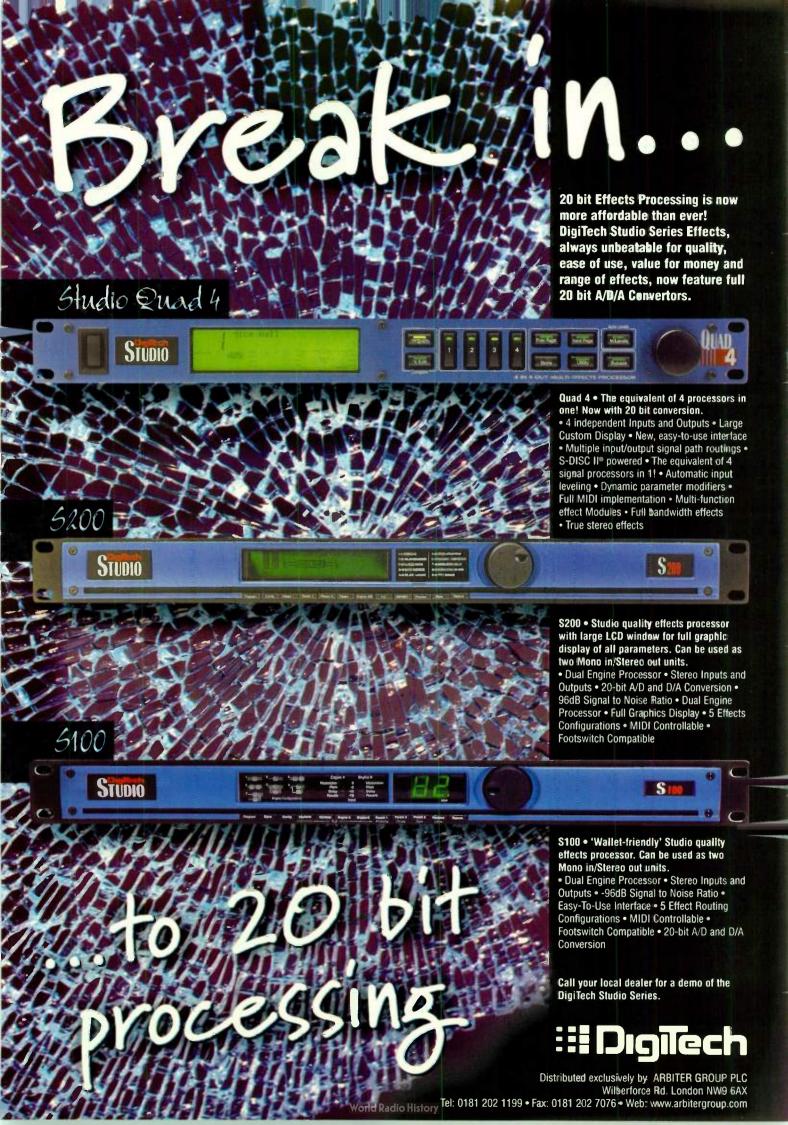
FX Rentals are having an novel competition: until the end of August, all account customers will be issued with a numbered plastic duck for every £200 spent. There will probably be around 3000 ducks, and on September 18, they will be raced along a stretch of river in Staines. The winning duck will win a complete refund of that customer's August bill, or a £2000 refund, whichever is greater. Second and third prizes, plus six runner-up prizes, will also be available.
T +44 (0)181 746 2121.

The final version of Creamware's SCOPE system, an open platform for sound creation, signal processing and studio integration, will actually be even more powerful than believed when it was shown for the first time at February's Frankfurt music fair. A new model of the SHARC DSP chip used by the SCOPE is 50% faster than that used on preliminary systems, and there will be 15, rather than six DSP units on the base-board. There are implications for the total number of in/out channels as well: each SCOPE card is capable of 64 ins and outs. SCOPE is due in October, with a retail price in the vicinity of £4000.
T System Solutions +44 (0)181 693 3355.
W www.creamware.de

Break in...

...to 20 bit processing

20 bit Effects Processing is now more affordable than ever! DigiTech Studio Series Effects, always unbeatable for quality, ease of use, value for money and range of effects, now feature full 20 bit A/D/A Convertors.

Studio Quad 4

Quad 4 • The equivalent of 4 processors in one! Now with 20 bit conversion.
• 4 independent Inputs and Outputs • Large Custom Display • New, easy-to-use interface • Multiple input/output signal path routings • S-DISC II® powered • The equivalent of 4 signal processors in 1! • Automatic input leveling • Dynamic parameter modifiers • Full MIDI implementation • Multi-function effect Modules • Full bandwidth effects • True stereo effects

S200

S200 • Studio quality effects processor with large LCD window for full graphic display of all parameters. Can be used as two Mono in/Stereo out units.
• Dual Engine Processor • Stereo Inputs and Outputs • 20-bit A/D and D/A Conversion • 96dB Signal to Noise Ratio • Dual Engine Processor • Full Graphics Display • 5 Effects Configurations • MIDI Controllable • Footswitch Compatible

S100

S100 • 'Wallet-friendly' Studio quality effects processor. Can be used as two Mono in/Stereo out units.
• Dual Engine Processor • Stereo Inputs and Outputs • -96dB Signal to Noise Ratio • Easy-To-Use Interface • 5 Effect Routing Configurations • MIDI Controllable • Footswitch Compatible • 20-bit A/D and D/A Conversion

Call your local dealer for a demo of the DigiTech Studio Series.

::: DigiTech

Distributed exclusively by ARBITER GROUP PLC
Wilberforce Rd. London NW9 6AX
Tel: 0181 202 1199 • Fax: 0181 202 7076 • Web: www.arbitergroup.com

World Radio History

shape of things to come

It spins you right round...

The software emulation of analogue soundmakers, as typified by Steinberg/Propellerheads' *ReBirth RB338*, is a pretty common technique now. But how about the simulation of an analogue front end, to control MIDI instruments? This is the direction taken by Canadian company Dimension Arc

The 103 Bassline monosynth modules emulate a monophonic analogue synth, but with a pattern sequencer to store/loop step-sequenced notes. You can control any parameter you want during playback, with six continuous controller knobs.

The 109 Drummer module emulates a classic analogue rhythm machine with seven

Software, with their *Revolve 100M* groove sequencer. To a certain extent, *Revolve* is compatible with *ReBirth* — it can play alongside it, and MIDI files can be transferred between them. However, *Revolve* generates solely MIDI data.

It's a pattern-based performance sequencer with analogue-style controls, user customisability, and a full MIDI spec: all on-screen controls send or can respond to MIDI controllers. You could use a MIDI keyboard to edit each step, the computer keyboard for general parameters, and MIDI faders to tweak the results. The basic v1.0 *Revolve* contains three MIDI sequencing modules: 103a Bassline 1, 103b Bassline 2, and 109 Drummer. The "pro" version — currently under development — will include additional sequencing and MIDI processing modules.

user-definable drum notes (assign them to trigger drum sounds, samples, or loops) and seven continuous controller knobs assignable to any parameter.

Currently, *Revolve 100M* is Windows only, and requires a minimum of a 486/66, 10Mb RAM, multitimbral MIDI sound source or sound card, MIDI interface, and a 640X480 monitor, 256 colors minimum. A Mac version is being considered. UK distribution has yet to be finalised, but a beta version is available from the Dimension Arc website, and details of ordering the full commercial version are also available online.

A Dimension Arc Software, #499 - 425 Carrall St, Vancouver, BC, Canada V6B 6E3.
T +1 604 664 0403.
E revolve@dimensionarc.com
W www.dimensionarc.com

Alesis ADAT

Type II ADAT Recorder

PAUL WHITE meets the new junior member of Alesis' family of Type II 20-bit ADAT recorders.

With the arrival of the Alesis ADAT Type II 20-bit recording format came the announcement that a new, lower cost ADAT would be joining the range for the benefit of those users who wanted the sound quality of the Type II format but not necessarily all the features of the XT20. The XT20, reviewed in *SOS* in May this year, has balanced analogue I/O and a comprehensive locator system; on the LX20, the locator system has been considerably simplified, and the analogue I/O is via unbalanced phonos only. However, the standard ADAT digital I/O remains, as do the nine-pin sync sockets to allow synchronisation with other ADATs. The LX20 has its own small remote control or LRC which is included with the package, as is one ADAT S-VHS tape, though full

only push them hard enough to get to the first click, nothing happens. Give them a slightly more decisive prod and they work as expected.

While the transport and audio converter circuitry is the same as that used in other Type II machines, the mechanical construction is much simpler than usual, relying on a folded steel chassis rather than on the heavy casting of the XT. The overall circuitry is also far simpler and tidier than on early ADATs, with excellent access to the transport for cleaning and servicing. The metal top cover is damped with a small square of acoustic foam but, in practice, it still rings quite noticeably when you tap it, and the transport noise is more evident than would ideally be the case. It isn't too bad in play or record modes (at least compared with computer systems), but fast wind is pretty industrial-sounding. Given that there's no meter bridge function on any of the new ADATs, I feel that mechanical noise could have been better addressed as it's invariably necessary to have the machines close by so you can see the meters.

As far as I can tell, the display is identical to that of the XT20 and provides large plasma meters as well as parameter and location

fly, and as well as being able to shuttle between locate points, it's also possible to set Auto Return and Auto Play to make the machine loop between locate points one and four. Auto punch in and out can be implemented between locators two and three with the option of a rehearse mode. For those less sure of their musical success, it's also possible to set up a loop record mode where you can go over the same take time and time again until you get it right. I don't know if it's just me, but I've never used an auto-punch

compatibility with the BRC controller/locator/synchronizer is also maintained, as is the universal 90 to 250 Volt power supply.

The front panel of the LX20 is a little simpler than that of the XT20, with no further row of buttons between the display and the track arming buttons. Furthermore, all the buttons are moulded rubber, giving the whole front panel a less 3D appearance than the XT ADATs. These new buttons appear to have an almost two-stage click operation, so if you

information. Function status LEDs are adjacent to the buttons rather than being moulded into the buttons themselves, but the layout is still very clear and easy to navigate around.

A maximum of four location points (in addition to return to zero), may be stored when the machine is being used in a stand-alone mode, though if it's being controlled by a BRC or slaved to XTs, it obviously obeys all their autolocate commands, which is why it makes such an attractive slave machine. Locate points can be dialled in by location or stored on the

mode in my life — I always find it quicker and easier to punch in and out manually, or with a footswitch connected to the BRC if I'm playing guitar at the same time. With the LX20, the LRC socket doubles as a regular footswitch input for punching in and out.

As with other ADATs, the digital crossfade used at punch in and out points to avoid clicks can be adjusted, in this case from 11 to 43 milliseconds. Once again, I've always found the factory default to be perfectly satisfactory. Pitch control of -300/+100 cents is available.

LX20

and if you want to know how many hours your machine has spent running, holding Set Locate and pressing Stop at the same time will tell you how long the drum has been spinning in contact with the tape. The same track copy and track delay features are provided as with other ADATs.

Instructions are provided for cleaning, though if good quality tapes are used, ADATs will run for a long time without problems. There's an inbuilt routine for recalibrating the brakes, which is suggested for every 250 hours, and an error 'asterisk' in the display comes on whenever interpolation is used to conceal an error. If this lights up regularly, try a different tape, and if that doesn't cure the problem, a proper cleaning will usually do the trick unless the machine is in need of a service.

USING THE LX20

Operating the LX20 is much like any other ADAT except that before you start work, you have to decide whether to format your tape to the old ADAT Type I standard or to the new 20-bit Type II format. Type I is 16-bit only, but will play back on both old and new machines. Type II offers the extra dynamic range of 20-bit recording, but can't be played back or overdubbed on older Type I machines, so the choice hinges on whether you are finishing the complete project yourself, or whether the tapes may need to be worked on at a facility that has only Type I machines.

The sound quality of the LX20 seems to be every bit as good as that of the XT20, but then as the converters and transport are the same, there's no reason to expect otherwise. Type II ADATs, including the LX20, are compatible with older ADATs for synchronisation purposes, providing the older ADATs have a suitably current software version. As with the original XTs, mixing them with 'black face' ADATs will slow down their wind and rewind operations as these older machines were considerably slower.
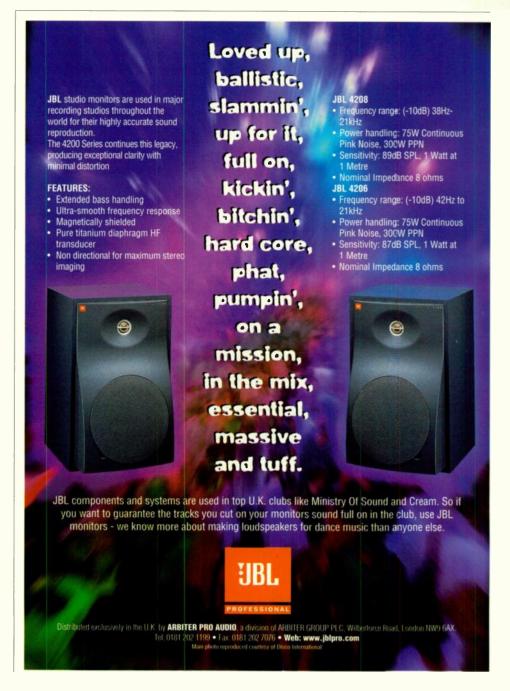
SUMMARY

ADAT LX20s will appeal to those users who want lower cost slaves to link up to a XT20, but the unbalanced analogue I/O may be seen as a limitation. It's OK in situations where the ADAT optical link is being used, but less satisfactory where you need to use analogue connections, especially when you consider that even the cheapest mixing consoles now tend to have balanced line inputs and group outputs. Other minor niggles include the

transport noise and the rather odd button response, but on the whole, the LX20 is extremely good news. Given that, for a significant cost saving, you still get the same 20-bit sound quality as from an XT20, the LX20 is worth considering for stand-alone use as well as for use as a slave, provided that you don't need the extra locate facilities.

The whole ADAT concept may seem a little outdated as more stand-alone digital workstations and software based studio packages appear, but the low cost and high storage capacity of tape has a lot to commend it. ADATs also provide a track-to-track sync accuracy that's to single sample resolution, while I've heard tales of people using software digital audio systems where the track sync errors have been so large as to be clearly audible. Until very high capacity, very low cost rewritable disk cartridges become a reality, and until computer-based systems are designed to be less problematic, I believe digital tape has an assured future.

Amtech Age One

Tape Echo Emulator

PAUL WHITE takes a trip back in time to a place where old technology is being used to emulate even older technology.

When the tape echo unit was first invented, it allowed musicians to approximate on stage what could previously only be done using large studio echo chambers. It didn't provide a perfect emulation, of course, and I've no doubt that musicians grumbled about it to start with, but they used it anyway, despite the risk of an embarrassing tape loop break in the middle of a crucial instrumental solo. Technically, the things were noisy, the repeat delays disintegrated into mud after a few trips around the feedback control, and you had to nurse them from one gig to the next, but oddly enough, they sounded really great when they were working properly.

In the '70s came the electronic echo box, based not on digital technology but on a then-new type of analogue chip called a charge-coupled device, or CCD. CCD delays worked by sampling the incoming signal in much the same way as

been lost, so the quantity of water you get at the end of the line isn't exactly the same as you started out with. A CCD chip does this using electrical charge passed from stage to stage, and the more stages of delay, the more inaccuracy (noise) creeps into the reconstructed signal at the other end. Because of the limited number of delay stages you could link up without suffering too much signal degradation, the sample rates were often quite low, resulting in a very limited audio bandwidth and audible aliasing distortion at the longest delay times. That meant the manufacturers had to build in compander noise-reduction chips to keep the noise to an acceptable level, but that lead to noise breathing, so although there was silence when you weren't playing, you could hear hiss mixed in with the delays as you played. Of course we all went out and bought these things, but soon realised they didn't capture the magic of a proper tape delay. However, there were no tapes to break, and the repeats became less clear, just as they did with tape!

Digital delay became affordable in the '80s, and eventually prices dropped to the point that even guitar pedal manufacturers abandoned CCD technology in favour of digital delays. This was it: clean delays with no loss of quality as the sound recirculated, very low noise and far longer delays than you could get with CCD technology. Great in theory, but it didn't sound

filtering, but have instead gone back to CCD technology and tried to bring it up to date. The result is a 1U rackmount box that purports to emulate the warm, unfocused sound of tape echo units such as those made by Watkins, Binson, Vox and those wonderful Miazzis that had tape echo built into their PA systems.

Configured as a mono-in, mono-out processor, with a choice of effect only or effect/dry mix outputs, the Age One has a high input impedance optimised, so I'm told, for use with Fender Strats, though it will of course work with any electric guitar. There are three sensitivity switch settings and an input level control along with a generous LED meter, but curiously, this only affects the delay level — the clean sound remains stoically unchanged.

digital circuitry does, but instead of representing it by numbers, they just sampled the instantaneous signal voltage and then passed that on to the charge-coupled device. CCDs are also known as 'bucket-brigade' delays, because the way they work can be represented by a long line of people with buckets. Imagine the sampled voltage is a quantity of water you want to pass to the end of the line. You pour it into the first bucket, then from the first to the second, the second to the third and so on. By the time it gets to the end of the line, some of the water may have

right — not only was it not as warm as tape, it wasn't as warm as CCD technology either! Hence the '90s syndrome of listening to old records by the likes of the Shadows and the Ventures, then asking how on earth they got that amazing guitar sound.

THE AMTECH AGE ONE

Amtech are a small Swedish company whose driving passion is to recreate the vintage tape echo sound using wholly electronic means. Interestingly though, they haven't used physical modelling, or digital delays with clever

A suitably retro-looking footswitch is included for bypassing the effect.

There are six switchable echo modes, the first three seeming to produce roughly evenly-spaced echoes and the remaining three designed to simulate multi-head tape echo boxes. The overall delay time can be varied up to a maximum of 450mS, and there's a delay tone control as well as adjustment over wet/dry mix and a feedback knob. There's no MIDI, no programmability and no multi-effects, but there is a knob on the back panel called Flutter, which attempts to emulate

the tape speed variations often associated with tape echo units that had spent too long between services. Rather than simply modulate the delay time rhythmically, this seems to have a random element to it, possibly produced by a noise-fed sample-and-hold circuit fed into an integrator. Whatever the methodology, the result is reasonably authentic with just enough range to prevent you getting silly with it.

BACK IN TIME

I tried the Age One with my trusty Strat (a red one at that), and pottered around with the unit for an hour or so trying to find out what it was best at and what it was worst at. Overall, the caricature of the old tape echo sound is actually very good, though at the longest delay times, you can just hear aliasing distortion creeping in, which is something tape echoes definitely didn't suffer from. There's also a noticeable amount of background noise, and though this isn't too serious for live work (and much less than most original tape units), it could be a problem in the studio where the guitar sound is very exposed. It's also possible to hear the compander working, so the noise effectively gets faded out when the delays

have finally died away, but in context, this isn't really noticeable.

The flutter control adds nicely to the vintage illusion, but what works best for me is the way the repeats become progressively less distinct as they circulate. I'm not quite convinced that the feedback delays have the same decay law as on a tape echo (possibly due to tape saturation?), but it's very close.
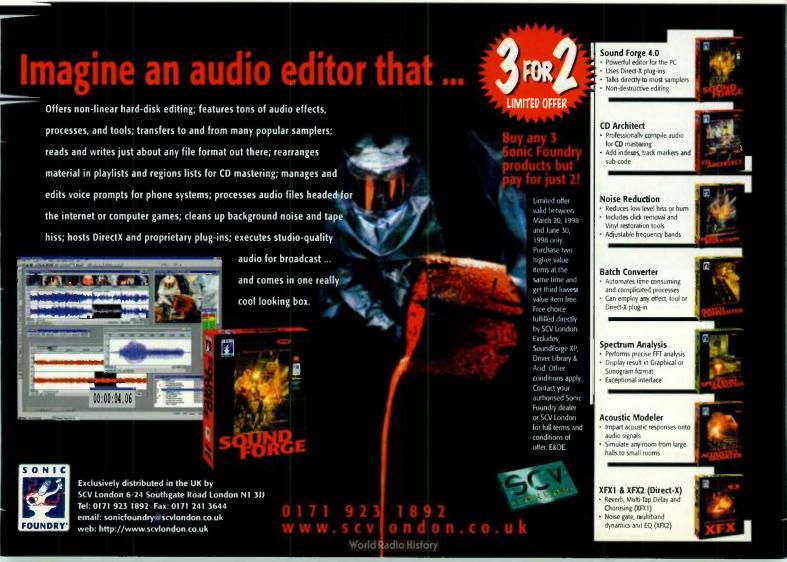
AGE CONCERN

The Age One is very nicely built, with hand-loomed cabling and quality components, and it must also take quite a long time to set up as the CCD chips are surrounded by the usual array of preset potentiometers, as are the compander stages. Unfortunately, when you combine labour-intensive assembly and expensive components with a niche market, the result is a high price, and at over £700, the Age One is certainly no impulse buy. I guess the main market for this product is probably Shadows tribute bands or Indie bands trying to recreate a '60s feel, and in that role, there's little else to recommend other than seeking out an original tape echo. I've owned a great number of CCD echo boxes and some came very close to sounding the way the Age One sounds, but having the multiple taps and the variable flutter adds to the authenticity. My own feeling is that

it wouldn't be that difficult to emulate tape echo digitally if someone had a mind to sit down and do it properly — while many people love the old sound, few feel equally nostalgic about the background noise that comes with it! However, until they do, the Age One stands alone, waving the banner for guitar nostalgia.

I think it's fair to say that at the asking price, the Age One is destined to remain a bit of a niche product, though the idea of a 'just add water' '60s sound will certainly appeal to some sectors of the music market. Whatever the economics of units like this one, it's refreshing to see small companies still doing something out of the ordinary rather than us all having to use the same pre-packaged, pre-digested sounds and effects, and if you're a Hank Marvin fan, I'm sure you'll love everything about it. **SOS**

Line 6 AX2 212

Modelling Guitar Amp Upgrade

Line 6 announce a software upgrade for their physical modelling guitar amplifier. PAUL WHITE plugs in, turns on and checks out the reborn AX2.

Back in June this year we reviewed one of the new Line 6 Flextone combos, not because we've suddenly become a guitar magazine, but because the amplifier uses state-of-the-art physical modelling and DSP technology to recreate vintage guitar sounds. This is obviously attractive for the player who wants access to lots of different types of sound, and because all Line 6 models include a speaker-simulated line output and a host of effects, recording from them is easy.

using the buttons, then turn the knob directly beneath to make adjustments. Apparently, even this was way too scary for some guitar players, hence the simplified Flextone models, but I was convinced enough to buy one. It sounded good, all the effects settings and levels were saved with a patch, and it was surprisingly loud. It was just what I always wanted, but last week I said goodbye to it forever. Why?

Line 6 have discontinued the AxSys 212 and replaced it with an even more powerful model, with better styling and a wider choice of amp and effect types. You'd expect this to leave existing AxSys owners somewhat out in the cold, but no — for just under £100, you can buy an upgrade kit that completely transforms your old amp into the brand-new Line 6 AX2 212 (alternatively, of course, you can buy the AX2 212 from scratch if you don't already have the older AxSys 212). The upgrade kit comprises a new stick-on front panel, an EPROM, a new keypad and restyled knobs. Though you can get a dealer to fit the upgrade, I found it very straightforward — the most complicated thing you need is a stubby cross-headed screwdriver. The self-adhesive front panel fits directly over the old one, and includes a complete listing of the amp models and effects types.

WHAT'S NEW?

Since the AxSys 212 was released, significant improvements have been made to the tube modelling,

What we haven't reviewed in *SOS* is the original Line 6 AxSys 212 combo; a heavy and powerful brute fitted with two 12-inch speakers, two 50W power amplifiers and a complete virtual rack of guitar amps, effects, fuzz pedals, wah and loads of other good stuff. Unlike the later Flextone models, the AxSys 212 featured a completely editable multi-effects section, plus more parameter tweaking in the amp models. Editing was easy — all the adjustable parameters were shown on a front-panel matrix with knobs at the bottom of each row and buttons at the start of each row. All you had to do was select a row

not only in tonality and responsiveness, but also in bringing together the various clean, crunch and dirty versions of individual amplifiers to form a single 'unified' model. The tone control arrangement has also been simplified so that the 'main row' physical knobs control the actual tonal elements within the model rather than

pros & cons

AXSYS AX2 UPGRADE £99

pros
- Gives the AxSys 212 a new lease of life, including a cosmetic upgrade.
- Additional amp and speaker cab models with numerous improvements.
- New effects and improved performance controls.

cons
- The amp still doesn't look as pretty as the new Flextone models.

summary

The AxSys 212 was a remarkable amplifier, and the ability to upgrade it as you would a computer or piece of software should ensure that it remains at the cutting edge of guitar technology.

SOUND ON SOUND

simply adding extra EQ on top of what was programmed in via the Preamp parameter row. You still get a post-model graphic EQ with a Presence control, but in order to stay true to the original amps, the Presence control is only available if the original amp had one. Similarly, the EQ position in the signal chain is as per the original amp, though when emulating amplifiers which only have bass and treble controls, the 'spare' mid EQ control is placed after the amp model to add a little more flexibility. This attention to detail carries through to other amp parameters, such as the Bright switch — again, available only if the original amp had one.

With the AxSys 212, the knobs had no effect until they passed through the previously programmed value, when they would 'pick up' control and work as normal. The new model can be set to work this way, but the default mode allows a knob to become active as soon

as it is moved. In this mode, regardless of what setting is programmed, the parameter jumps directly to the value indicated by the knob position as soon as the knob is moved slightly.

The original AxSys included models based on the analysis of a number of classic Fender, Marshall, Soldano, Boogie, Roland and Vox

amplifiers. That list has now been expanded with the addition of a 1960 Fender Champ, a 1952 Fender Deluxe, a 1964 Fender Deluxe Reverb, a 1965 Fender Twin (an improved version of the original Fender Twin model), a 1960 Vox AC15, a 1985 Mesa Boogie IIc ++, a 1988 Soldano preamp and two rather exclusive American hand-built amps, the Budda Twinmaster and the Dumble Overdrive Special. These new amp models are all unified, so you don't have to switch models to get clean or dirty sounds.

To make the effects section more exciting, there's now an Octave Box based on the Tychobrahe Octavia, a tape echo simulation where the repeats get muddier as they die away, vibrato, and a Ring Modulator — useful in experimental music. Users with a Line 6 Floorboard controller will also notice an improved wah-wah, and the boost switch has been changed around to provide a real gain boost over and above that set up for the amp model.

There are further cabinet emulations, so that your choice of amp model can play through one of more than a dozen cabinet types ranging from an open back 1 x 12 to a closed 4 x 12 with speakers from either side of the Atlantic.

MAKING THE CHANGE

Once the upgrade is installed, all the existing user patches are translated 'on the fly' to work with the upgrade. Existing patches will undergo some slight tonal changes due to the revised models used in the upgraded amp, most of which are slightly brighter than those of the original. It's not until you physically save the patches again that they are stored in AX2 format — until then they are still AxSys patches. The factory patches also change when you upgrade, and now the banks (each bank contains four sounds) tend to be based around specific amplifier types. For example, Bank 1 is based on the Roland JC-120, Bank 2 on the Fender Deluxe and Fender Champ, Bank 3, Fender Bassman and so on. Banks 20 to 22 feature specific effects setups while the remaining 10 banks are modelled on famous players. You get Jimi Hendrix *Little Wing* and *Purple Haze* tones, Eric Clapton circa Cream, Dave Gilmour, Mark Knopfler, BB King and so on. It must be stressed that these patches aren't created by, or endorsed by the artistes themselves — they're simply put together by the guys at Line 6 to show what their amp can do, and all will require some adjustment if you don't have the same guitar (or playing chops!) as the original artist.

The AX2 is a great new amp for anyone looking to get all the guitar sounds and effects they need from one box, and though a real tube amp feels a little different to play, I've found the tonal flexibility to be worth any tradeoff. This certainly comes closer to the elusive valve sound than anything I've tried that didn't actually contain valves, though players who prefer to use just one or two sounds will probably be happier sticking with their favourite amp.

For the existing AxSys 212 owner, the upgrade is a real bargain — not only does it freshen up the appearance of your amp, it also greatly adds to its tonal flexibility. The tape delay is a particularly nice touch, as is the Fender Champ model — my other amp is a Fender Champ, and this one can get pretty close. The bottom line is that this upgrade makes what was already a worthy studio amplifier even better, and for gigs, it certainly saves a lot of time wiring up effects and matching levels. **SOS**

ART Tube Pac &

Preamp/Compressor & EQ

PAUL WHITE tests ART's two new compact tube processors to see if they sound bigger than they look.

When ART first introduced their Vactrol-based opto-compressor a couple of years back, I was more than pleasantly surprised by its sound, so when they announced they were combining a simplified version of this compressor with one of their tube mic preamps, I was keen to try it. As fortune would have it, they also have the Tube EQ in the same series, which arrived at the same time, so I got to play with that as well.

similar meter directly below to show the amount of gain reduction taking place, up to a maximum indicated value of 12dB.

I don't think I've ever seen a simpler compressor section than this one — but don't let that put you off, as it actually sounds pretty good on most material. There's a variable threshold control and an overall output gain control for the unit, but other than that, there are just four buttons, if you include Bypass and the Phase Invert switch. Compression is of the soft-knee type, with a choice of two ratio settings labelled Comp and Limit — Comp has a ratio of 2.3:1 and Limit 6:1. The attack time is fixed at around six milliseconds, and the release time may either be fast or programme-dependent, again selectable via a switch.

As far as I can tell, this unit uses the same low-voltage tube circuitry as its predecessors, so the overdrive characteristics may not sound

range of the signal with very few audible signs of compression, but for a more overtly compressed sound, adjusting the threshold to give 12dB or more of gain reduction will do the trick. Conversely, the Limit setting (6:1 ratio) can be used to give a much more 'squashed' sound, but because of the soft-knee compression curve, the less gain reduction you set, the less obviously compressed the sound seems. At quite high gain reduction settings, the Limit mode is pretty unsubtle, but still in a smooth and musical way.

The Tube EQ.

Both devices come in a 1U-high, half-width format, and as their hybrid circuitry includes tubes, they don't inflict further wall-wart PSUs on us, but are mains powered (hooray!). The casings are both tough and attractive, with nicely curved front panels, and the white-on-black legending is clear and easy to read.

Taking the Preamp/Compressor first, this combines a mic/line preamp with an opto-compressor where each section includes its own ECC83/12AX7 tube. Input is via a balanced XLR for mic-level signals or a high-impedance input for line-level or guitar signals. Switchable 48V phantom power is fitted, along with a 20dB gain boost stage and phase inversion (the latter following the compressor stage). Outputs are available on both unbalanced jack and balanced XLR, so the unit may be used either as a direct recording channel or patched via a console insert point. A small, but effective 4-LED meter shows the degree of tube drive, with a

The Tube PAC.

exactly the same as a high-voltage design. Nevertheless, this type of circuit can still sound good, and it helps keep the price down.

THE SOUND

The circuitry may use low tube voltages, but it still sounds impressively clean, providing you don't drive it too hard (some 'starved tube' circuits can sound rather soggy). Background noise level is reassuringly low; my Rode NT1 mic sounded very sweet indeed.

At the lower compressor settings, it's possible to knock 6dB or more off the dynamic

Trying a guitar through the high-impedance jack input provides a nice combination of DI and compression, again with minimal noise. The preset attack of the compressor works nicely with guitar to give the attack of the sound a little life, but it isn't so pronounced that it sounds unnatural on vocals.

TUBE EQ

Packaged in the same case as the Preamp/Compressor, the Tube EQ has independent input and output level controls

Tube EQ

and a choice of balanced XLR or unbalanced jack connectors, but there's no mic preamp in this model. The front panel proclaims this a parametric equaliser, but it's not. as there are no Q (bandwidth) controls — it's actually a 4-band EQ with two sweep mids and shelving high and low sections. The shelving frequencies can be switched from 40Hz/120Hz and 6kHz/18kHz respectively, but the mids have no Q adjustment, not even a two-position switch setting.

Though they're not parametric, the mids do feature switchable frequency ranges as well as +/-12dB of gain: Lo-Mid goes from 20Hz to 200Hz with a 'x10' switch to take it up to 2kHz, whereas the Hi-Mid starts at 200Hz and goes up to 2kHz with the 'x10' switch pushing it up to 20kHz. This leaves a useful amount of overlap between the ranges to ensure you can get at exactly the areas of the spectrum you want to. Both Mid controls use dual-concentric pots and knobs to save panel space, while a single Bypass button takes the whole EQ out of circuit.

TUBE SOUND
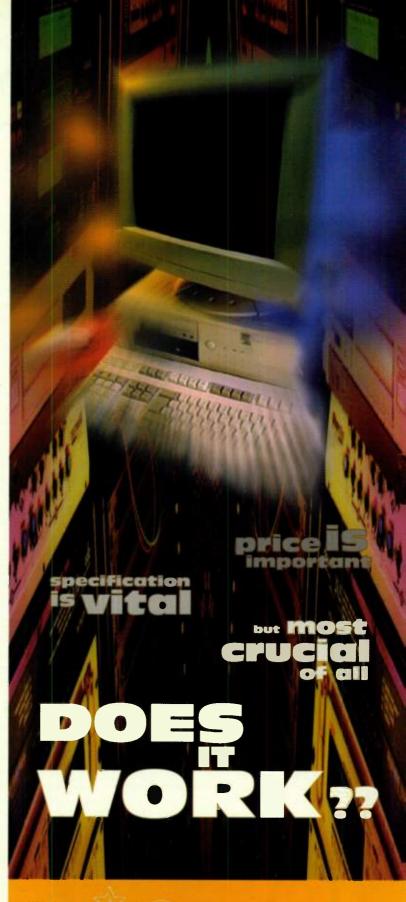
It turns out that the Tube EQ is well-behaved and reasonably sweet-sounding, though it doesn't have the flexibility of a parametric equaliser, especially when you need to home in on a narrow part of the spectrum. Even so, it has a wider range — and is rather nicer-sounding — than many console EQ sections. The switchable shelving filters add to the usefulness of this equaliser, and in many situations, the lack of fully parametric mids may not be that important. as the filter widths have been sensibly chosen.

SUMMARY

Both these units are well built, affordable, and have a clean, professional sound augmented by just a little tube warmth. For me, the Preamp/Compressor is the most attractive, as it offers mic/line/instrument preamp facilies as well as its very smooth-sounding compressor section. The Tube EQ also sounds very good. but the lack of true parametric operation or a mic amp makes it less of an all-rounder. I suppose ART would like you to buy both and patch them together!

While esoteric tube processors are nice for those who can afford them, ART provide a very valid 'halfway-house' product that improves on the facilities found in a typical mid-price mixing console, yet costs far less than the top-end tube models from the big names. There are already a number of worthy products in this area (both hybrid and all solid-state), including the excellent Joemeek range and TL Audio's hybrid processors, so it pays to listen to a few different models before deciding which one suits you best. You may also be surprised at how little tonal difference there is between some of the solid-state and hybrid tube models, so go by what your ears tell you, not just the spec sheet. Whatever you do though, don't leave these two new ART models off your shortlist!

SOS

Sony SMS1P

Powered Monitors

PAUL WHITE tests Sony's diminutive powered monitor designed for desktop music and multimedia work. Can such a small box deliver the goods, and is the performance worth the price?

When I first became interested in recording, the nearfield monitor as we know it today didn't exist — instead, perched on top of the desk would be a pair of Auratones. These old Auratones had three distinguishing points: they were tiny, they were inexpensive, and they could go very loud. You'll notice that none of the three points relates in any way to their musical accuracy! The Auratone used a single driver, around four inches in diameter, mounted in a sealed wooden cube just big enough to contain the driver. It tended to be middle heavy, so whenever you played a mix back on anything else you'd find the vocals had dropped in level by around 3dB.

The reason for mentioning the original Auratone is that Sony have gone for the same single driver approach in designing their little desktop powered monitor. Unlike Auratones, however, the SMS1Ps are active, they're ported to extend the bass and they're magnetically shielded.

AT FIRST GLANCE

Measuring 132 x 210 x 230mm, the cabinet contains a single three-inch driver augmented by a frontal bass port and four control knobs. The SMS1P has two separate, mixable audio inputs, each with its own volume control. One channel has a choice of balanced XLR or mono jack inputs, while the other offers jack or phono. Mains comes in via an IEC cable and the internal amplifiers are rated at 15 Watts. The remaining two rotary controls provide bass and treble control over a plus or minus 6dB range — there are no centre detents on the controls and no EQ bypass. If you look at the frequency response, it could only be described as ruler-flat if the ruler in question had suffered an accident involving a combine harvester and a set of crimping irons! Even Sony could only bring themselves to quote the frequency response as 80Hz to 16kHz (-10dB) rather than the usual -3dB points. However, and here's the ironic thing, the SMS1Ps actually sound pretty good!

Top speaker designers stress the importance of smooth phase response, and in doing away with multiple drivers plus a crossover, Sony have neatly removed a major source of phase anomalies. The tradeoff, of course, is that high frequencies will tend to roll off earlier and the dispersion angle is bound to narrow as the frequency increases, which means a narrower sweet spot and a less bright contribution to the overall sound from room reflections. Nevertheless, if you're working close to the monitors, this need not be too serious a problem.

HEARING IS BELIEVING

Checked out with my standard test discs, the SMS1Ps sound surprisingly uncoloured and well balanced, though there is a slight lack of top-end crispness. Nevertheless, cymbal sounds come over cleanly, and you get to hear the kind of information in a mix that a more complex monitoring system will occasionally obscure. At the low end, there's enough punch to get a convincing kick drum sound and vocals are handled particularly kindly — not entirely unexpected as most other monitors' crossover frequency is right in the middle of the vocal range. What's more, the vocal balance seems pretty natural. There's no quoted maximum SPL, but I noticed the speakers quickly changed from sounding neutral to sounding boxy if worked too hard, and the maximum clean level isn't

over-generous. There's enough level for recording and mixing with the speakers mounted at around arms' length, but any further away and you may find the lack of power frustrating.

SUMMARY

Despite my initial reservations, the SMS1Ps actually sound pretty good, though I'm not convinced that their performance justifies their almost £400-per-pair price tag. They have a smooth, non-fatiguing sound that suffers only slightly from lack of detail through having no separate tweeter, and overall, they deliver a well-balanced impression of the mix you're listening to. On complex material, the sound can become a little confused, and the SMS1Ps are let down slightly by being somewhat limited in the maximum SPL they can deliver. For their intended desktop applications, though, they go just about loud enough, and the magnetic screening means you can use the speakers closer to VDUs. However, there are plenty of alternative monitors to choose from in the same price range (even if you add the cost of a passive monitor to the cost of a suitable hi-fi power amp), and a significant number of them would outperform the SMS1Ps in both fidelity and overall power. **SOS**

Yamaha 01v In Stock

The latest in an illustrious line of best-selling digital mixers. The 01V is the ideal digital mixer for project studios, broadcast and live sound, providing enormous flexibility and features. The 01V has 24 input channels, 8 optional digital inputs, 4 assignable analogue outputs, EQ, dynamics and 2 stereo multi-FX processors, automated mixing and instant recall. 60mm motorised faders, a large backlit LCD and a huge onboard library of EQ and Dynamic processor presets make using the 01V a breeze. Link two together for an incredible 48-input console at an even more incredible price.

Promix 01 - 1 Only £899

Tascam DA20MkII

Industry Standard DAT recorder now only **£489 inc** 5 FREE DAT tapes.

Nord Modular
(PC/Mac)
Incredible modular sound creation & editing on your Mac/PC

£Best UK

All Behringer mixers and effects in stock at the lowest possible UK prices

Package deals available

BEHRINGER

DEALS!

Fostex FD4

Incredible new low cost all digital multi-tracker from Fostex. There's room for a 2.5" internal drive and SCSI as standard. An integral mixer and MTC sync capabilities make this a perfect introduction to the Audio and MIDI environment. **£Best UK**

deals with Zip or EZ Drives available

Fostex DMT8v In Stock **£799**

Phat-Boy
Realtime controller to bring your synths *ALIVE!* **£139.95**

V2

Yamaha A3000

The new version 2 A3000 offers expandability to 128Mb with a whopping 64-note polyphony, loading WAV, AIFF and AKAI formats and save as AIFF. A new Loop Remix function for recycling your beats, Loop Divide slices up beats and maps them across your keyboard.
16 filter types, REALTIME pitchshift, ten outputs and SCSI as standard **£Best UK Deals**

Event Layla

IN STOCK

Call now for the best UK Sales & Support for this world beating system. Special discount prices on package deals...Computer systems set up and ready to go. Call for details.
Gina In stock
Darla In stock

All Turtle Beach cards and upgrades in stock

Montego £call
Fiji £low
Pinnacle £low

MICROPHONES

Amazing Deals on...
RØDE AKG audio technica
Special Offer: C1000s £129

MIDI & Audio

CAKEWALK

Cakewalk V7.0 is here - the slickest fastest most powerful Cakewalk yet! **£Best Uk**

emagic

Logic family gets bigger and better with direct-X and unprecedented audio channel support. Call for updates. **£Best Uk**

CUBASE

Cubase VST v4 introduces Steinberg's most powerful audio and MIDI sequencer to date. (Mac Only) **£Best Uk**

ADAT digital i/o

SONORUS

KORG Wave Center

The ADAT standard enables the transfer of 8 digital audio channels down a single optical cable, so you can connect your digital output from your 8 bus mixer to the input of your computer, and vice versa. Here are our 3 best selling ADAT cards.
Korg 1212i/o
ADAT i/o plus wordclock and ADAT sync **£Call**
Sonorus Studi/o
2 x ADAT i/o with optional ADAT sync **£New low price**
Frontier Wave Centre
ADAT i/o plus S/PDIF & 1 MIDI in 3 MIDI Out **£399**

FAST Fast Multimedia AG
miro

DIGITAL VIDEO

Call Mike for details of our digital video editing packages. We stock all the leading names in desk-top post prod.

If you've £250 to spare and a reverb-shaped hole in your studio, Lexicon would like you to fill it with their latest budget processor, which offers 20-bit converters, a digital output as standard, and the famous Lexicon Ambience effect. PAUL WHITE weighs in with his balanced opinion...

The MPX100 is the first in a range of processors based on Lexicon's new Lexichip III reverb processing engine, an extremely powerful custom device capable of running Lexicon's famous high-quality reverb algorithms without cutting corners. Lexicon have ventured into the budget effects market before, notably with the Alex and the Reflex, so affordable Lexicon effects are no longer unheard of, but the MPX100 offers a few refinements that could make it attractive to anyone searching for decent-quality effects on a shoestring. For a start, it differs from its predecessors in that it has 20-bit input and output converters running at 44.1kHz. Combined with an internal 24-bit signal path, this gives a dynamic range of over 95dB using the analogue output, but Lexicon have also included an S/PDIF digital output that extends the dynamic range to better than 100dB.

Lexicon's new unit provides reverb, as you'd expect, but also adds a wide selection of non-reverb effects, from delay and chorus to rotary speaker simulation and pitch shift. The control system hints at the well-worn Alesis Microverb/Midiverb

Tap time, as part of a user patch. Tap is used to set the delay time of certain echo-type effects or the pre-delay time of some reverb patches. It's also used in various presets to set modulation rates.

Even the Adjust knob isn't as straightforward as it seems —in some modes it operates as a conventional 'amount' knob, in others it operates either way from the centre position, and sometimes it behaves as a switch. It also alters more than one parameter on some patches, a trick it doubtless learned from the PCM80 and PCM90! There's a Store button for saving edited patches back into one of just 16 user memories, and holding this down at the same time as Tap puts the box into MIDI learn mode, so that an external MIDI controller can be used to modulate a key effect parameter.

EFFECTS

Both single and dual effects are featured on the MPX100. All the single effects are available on the left-hand side of the rotary Program Selector switch, with the 16 variations of each selected via the Variation knob. As you'd expect, the usual reverbs, choruses, flangers and delays are there, but — for the first time on a budget Lexicon unit — you also get their classic Ambience algorithm, which is great for making a dry sound 'belong' in a mix without adding obvious reverb. On top of that,

a ton better?

LEXICON MPX100 DUAL-CHANNEL EFFECTS PROCESSOR

arrangement, where the user simply picks one of 16 variations on one of 16 preset effects themes, but there's actually rather more to it than that.
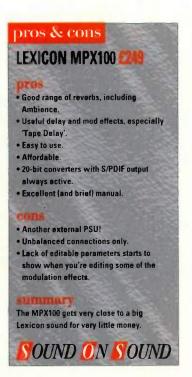
HARDWARE OVERVIEW

Physically, the MPX100 is a 1U-high, 2-in, 2-out effects processor based on 240 presets that can be edited on a fairly basic, but nevertheless useful level. The analogue I/O is on unbalanced jacks, with a high-impedance input that will accept electric guitars as well as regular line sources. For practice purposes, the left output jack doubles as a phones output, though it's not very powerful.

The colour scheme is traditional Lexicon and the stylish knobs have a very up-market feel to them. There's no display window, though, and only dual red/green LED indicators to monitor the input level. For the benefit of live players, there's a rear-panel Tap and Bypass footswitch socket. Power comes from an external adaptor.

A surprising amount of MIDI functionality is built into the MPX100, not least the ability to synchronise certain delay and modulation parameters to MIDI Clock. It's also possible to access all the patches via MIDI, using Bank Change and Program Change messages. Naturally, there are two MIDI sockets for MIDI In and MIDI Out/Thru.

Conventional rotary controls are used for setting input level, wet/dry mix and output level, and all but the input level setting can be stored, along with the

there's a selection of tremolo effects plus a rotary speaker simulation where the two virtual rotors spin independently and in opposite directions. The braking and acceleration rate of the real thing is also carefully modelled. A surprisingly sophisticated pitch-shifter is also included.

In the delay section, long delays of up to 5.7 seconds are possible, and the delays all have a choice of clean or tape-style operation, the latter producing more diffuse and warmer-sounding repeats. These are both available within each preset by turning the Adjust knob clockwise from centre for tape echo and anti-clockwise from centre for regular, clean DDL effects. Many of the modulation effects feature up to six voices (a spin-off from the PCM80), so you can create very rich chorus effects.

The dual effects patches combine various modulation effects with either delay or reverb, and there's also a 'delay plus reverb' configuration. It's in this mode that the Adjust knob operation sometimes gets a bit clever, because it adjusts the contribution of the two effects within the current routing setup by controlling how much of the dry signal and how much of the first effect is fed into the second effect (at extreme settings, the result is one effect or the other). Unless you have a trainspotter memory you'll need the manual while editing, as the routing is different depending on which variation you choose.

The same system is carried through all the dual effects, such that the first six variations sum both inputs to both effects blocks, which are arranged in parallel. The two sets of stereo outputs are then mixed. The next four variations are based on two stereo-in, stereo-out effect blocks in series, while the following three are true dual effects, arranged so that the left input feeds one effect and the right

input the other. Again, the stereo outputs are summed to provide a composite stereo signal. Finally, there are three further variations where each input feeds its own mono-in, mono-out effects block. Most of these combinations have editable parameters controlled by both the Adjust knob and the Tap button.
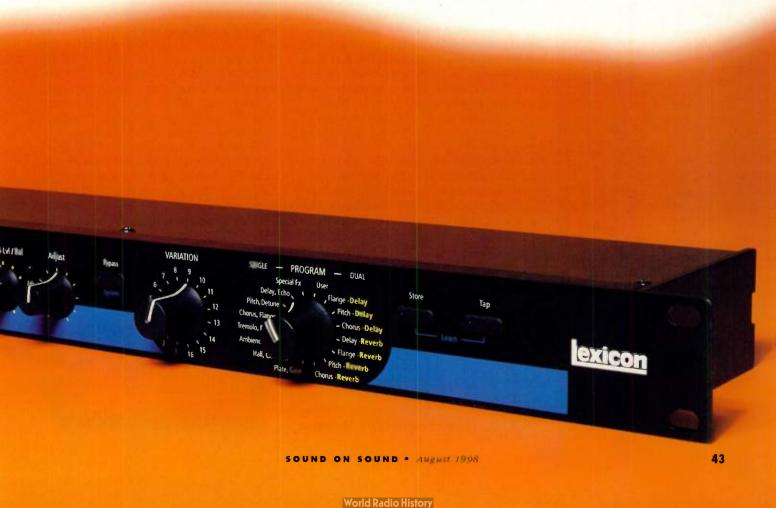
One bank falls into neither the dual or single categories, as it produces special treatments that draw on elements from other effects. Here you'll find an infinite reverb (actually, it's stopped now, but it's good for a minute or so!), a weird combination of pitch-bend and delay called The Abyss, ducking reverbs and delays, spiralling pitch shifts, infinite repeats and panning delays.

IN USE

Operating the MPX100 is extremely simple, but without the manual you might be a little lost as to what the Adjust knob, or even the Tap button, is controlling. The overall sound quality is much better than you'd expect from a budget unit and the input is reasonably tolerant of small overloads. Patches with lots of feedback, such as flanging, tend to be just a little noisy, and the large amounts of feedback make them vulnerable to overload, so you have to be rather careful with the levels. This is true of all effects units, but an analogue input limiter, such as the one on Lexicon's own MPX1, might have made setting up easier.

As for the sound of the new unit, the reverbs have the classic Lexicon character that allows you to use very high levels of effect without drowning the dry signal. In terms of quality, they're a lot closer to the MPX1 and PCM80 than they are to the old Reflex and Alex, and all have a good sense of 'realness'. In most cases, the Adjust knob sets ▶

"For the first time on a budget Lexicon unit you get the classic Ambience algorithm, which is great for making a dry sound 'belong' in a mix without adding obvious reverb."

▶ the decay time or tonality of the reverb and Tap, where applicable, varies either pre-delay or an echo delay. Some patches also have pre-delay mapped to tempo, so that when MIDI Clock is running you get a 32nd-note pre-delay.

The reverb algorithms on some reverb units can sound rather 'samey' from one to the next, but with Lexicon each algorithm is distinctive. They're very three-dimensional effects — even the smaller rooms, which have exactly the right character and coloration. What's more, most of the algorithms are suited to music production, so as well as typical concert halls and churches, you get bright rooms suitable for drums or guitar, intimate live rooms, vocal rooms, and so on. Especially interesting is Ambience, an algorithm first used on the big, expensive Lexicon 300s and 480s, and later on the PCM90 and MPX1. This emulates the reflections and colorations imposed by a real space, so it's good for placing dry-miked sounds in a real space without having to add reverb. The smallest ambience is designed for voice-over work, and the treated sound seems almost deader than the dry sound, if that's possible. At the top of the ambience scale is Marble Foyer, where the early reflections are quite obvious, but still die away quickly. Here the Adjust knob controls 'liveness', making this a useful preset for drums and acoustic guitar.

In the tremolo section are several different modulating waveform shapes, ranging from a choppy square wave to a rectified sine wave reminiscent of old guitar amps. Stereo pans are included, with various phase relationships between the left and right channels, as well as a handful of rotary speaker effects. The latter are musically pleasing and have the right speed dynamics, but they still don't seem quite real.

The ability to specify multiple voices on the MPX100 produces a nice rich chorus effect, but in most cases I prefer the more subtle detuning available from the Pitch effect repertoire. For more radical pitch effects, the shift can be set from -2 to +1 octave, and although the processing delay is a little irritating sometimes, the effect itself is a lot smoother than on most of its competitors. All the dual effects worked smoothly for me, with the MPX100 making no apparent compromise on effect quality as a result of generating two effects at once.

SUMMARY

It's very easy when reviewing a product like this to forget the price and start listing compromises, but at the asking price of £250 there's nothing I can't forgive the MPX100. OK, it is a little annoying to have to pick up the manual when programming, there are only 16 user memories, you can't name

IT'S ABOUT TIMING

Lexicon have made it very easy to synchronise delay and modulation parameters — an important issue, as even the best sounding delay is useless if it's running at the wrong speed for the track. On patches where the Tap function is available, a LED next to the Tap button illuminates, and if it's a delay preset, tapping the button twice will set a delay time equal to the spacing between taps. Of course, you can't get a longer delay than the maximum available in the preset you're editing.

If you're hooked up to a MIDI sequencer and want to change the tempo mid-song, you can assign a MIDI Program Change to the Tap button (you can also assign another to Bypass). A simple Auto Learn function means that once the MPX100 is primed, the next received Program Change is adopted as the controlling one. A similar Learn system is used to select a continuous controller to access the Mix, Effects, Level/Bal or Adjust knob functions. The exact parameter set by Tap varies from preset to preset, so you will need to keep the manual handy — sometimes it's a delay time, sometimes it's a reverb pre-delay, and sometimes it's a modulation rate. Another neat operational refinement entails holding down the Tap button to allow two audio input events to set the time. — for example, two drum beats or guitar plucks.

Sync is also possible over MIDI, and the MPX100 recognises MIDI Clocks in the range 40-400bpm. Whether or not the MPX100 syncs to incoming clock can be set up in the System mode, along with the MIDI receive channel and bypass mode. For bypass, there's a choice of either muting both the dry sound and the effect feed or leaving the dry sound active in bypass.

MPX100 PROGRAMS

Effect Type	Number of Programs
• Plate Reverb	8
• Gated Reverb	8
• Hall Reverb	8
• Chamber Reverb	8
• Ambience	8
• Room Reverb	8
• Tremolo	8
• Rotary	8
• Chorus	8
• Flange	8
• Pitch	8
• Detune	8
• Delay-Echo	16
• Flange-Delay	16
• Pitch-Delay	16
• Chorus-Delay	16
• Delay-Reverb	16
• Flange-Reverb	16
• Pitch-Reverb	16
• Chorus-Reverb	16
• Special FX	16

your patches, and there are not many editable parameters, but the vast number of presets means it's easy to get close to what you want very quickly, and the reverbs — which is what you generally buy a Lexicon for — sound way better than they've any right to do for a box of this price. In fact, I'd go so far as to say that most people wouldn't be able to tell whether they were listening to an MPX100 or the rather more expensive MPX1 in a mix. Having 20-bit converters and a digital output also helps ensure the MPX100 will remain compatible with your future setups, though the lack of balanced I/O may upset a few professionals.

As a main reverb processor, the MPX100 will more than satisfy most project studio requirements, in terms of both versatility and quality, and the fact that it offers other impressive effects too makes it a useful all-rounder. Those who've already got a decent main processor will still find the MPX100 a good second unit, and anyone without a Lexicon may be surprised by just how well those Lexicon reverbs sit in the mix. Frankly, I think other budget effects in this price range may find the going pretty tough from now on! [SOS]

THE CRAFT

CREATIVE SEQUENCER TIPS

PART 1: Your computer sequencer can do bewitching things with simple ingredients, creating the impression that you're a musical wizard even if you're just a one-finger pianist. NICHOLAS ROWLAND works some MIDI magic...

I think it was Brian Eno who was the first to put into words the idea that the recording studio was an instrument in its own right, and that you could learn to play it (in the sense of producing music or sound with it) in just the same way as you might learn the piano or violin.

I happen to feel the same way about sequencers. Although I'm interested in creating music — and have created hours and hours of it over the years — I'd be the first to admit that I'm not really a musician. What I do is use software tools to manipulate and edit MIDI data, the end result being a product that, on a good day with a following wind, you might just be able to hum along to. Some musicians might think this attitude is something to be ashamed of. But I've always felt that if you're interested in creating technically-based music, yet don't happen to possess the genius of Mozart, you should make full use of any help you can get.

So what follows is a collection of hands-on ideas and tips put together from my own experiences with sequencers. Like any article based on personal observations, it comes with the usual health warnings: first, these ideas are by no means exhaustive and second, be aware that the value of MIDI data can go down as well as up. The third thing to note is that these ideas are probably more applicable to riff- or loop-based styles of music (my own so-called .compositions. are in dance/ambient/techno territory) rather than formal songs as we know them, Jim.

COPY SHOP

What I've always liked about computer-based sequencers is that that they're able to do a lot of number-crunching very quickly, and also that the results are not necessarily what you might have predicted if you'd sat down and thought it out beforehand. So if you're feeling lazy (or technically inspired, depending on your point of view) you can use even the most basic of sequencer editing functions to make a relatively small amount of musical input go a long way — particularly if, like

mine, your music tends to be constructed building-block fashion. In other words, if you've started with a riff or melodic sequence, it's relatively easy to begin generating accompanying material simply by copying and pasting the basic material to new tracks and then using the sequencer's powers to manipulate it in various ways. To kick off with an extremely basic example:

- Copy your original sequence to a new track, then transpose it by an octave either way, or, if you're feeling more adventurous, a fifth or a seventh.
- If you choose a complementary or contrasting voice for the copied track, then pan each one to opposite sides of the stereo spectrum, you've already started to fill out your burgeoning arrangement — and you haven't even expended any energy in thinking up something new.

Of course, this is all kindergarten stuff. But you can take things further — for example:

- Double up a part and then delete alternate notes in each part.
- Pan the two tracks left and right to give you a ping-pong echo effect, which, again, may help to give some movement to your original sequence.
- A further variation on this trick is to copy your original riff to, say, four or five different tracks and assign them all to different versions of the same type of sound (a selection of pianos, guitars or organs, for example).
- Now delete notes such that your four or five tracks take it in turns to play each note — in other words, the first note of the sequence is played on the first track, the second on the second track, the third on the third and so on, until you come back to the first track again. You can now have fun introducing slight variations to the voices, perhaps in terms of their pan positions, their relative volumes, modulation or aftertouch.

If you're into dance music, this approach can provide an interesting way of building up and stripping down your basic sequence for intros, breaks and so on. Because you've effectively put individual notes on separate MIDI tracks, it's then easy enough to use the arrange page on most sequencers to mute different combinations of notes, creating new variations of your original sequence. You might even come up with a version which is actually more inspired than the original.

Doubling up tracks also takes you into the realm of MIDI-created effects. For example:

- Copy one track to another, then set both to play the same voice from your synth or sound module on the same MIDI channel. The result is a kind of phasing effect as each note double triggers.
- Those with an enquiring mind will be interested in taking this a stage further with triple or quadruple parts — the more MIDI notes you have trying to squeeze down the cables all at one, the more potential there is for weird glitches that can help add some zest to tracks. ▶

World Radio History

Cutting it Fine

THE SCIENCE OF DIGITAL AUDIO DATA REDUCTION

Though you might not realise it, the audio industry has employed data reduction strategies since the earliest days of digital systems. HUGH ROBJOHNS explains the concepts and explodes some myths.

You might think of data compression and reduction as a relatively recent development in digital audio circles — certainly it's only in the past few years that project studio musicians have started to become aware of data reduction techniques, various forms of which are employed in Minidisc recorders and some of the currently popular personal digital multitrackers to fit large amounts of audio data on to compact storage media without compromising perceived audio quality. But even in the Jurassic period of digital audio evolution, simple data reduction techniques were employed to avoid stretching the available recording and transmission media — techniques such as reduced sample rates and low-bit quantising. These days, data reduction is far more sophisticated, and allows us to achieve more with less — more channels, longer recording time, or better perceived resolution, with smaller discs, narrower tape, and restricted data-rate channels.

Although the process of reducing the amount of data needed to represent a digital audio signal is commonly referred to as **data compression**, the more accurate description is **data reduction**. Compression implies a reversible process (you can expand the compressed material to restore the original) but the majority of data reduction strategies are **lossy**, meaning that data is thrown away irretrievably. A **loss-less** system (and there are a few) is one where some data from the original signal is not recorded, but can nevertheless be recreated perfectly on replay — truly **data compression**.

Data reduction is not restricted only to digital audio; noise reduction processes such as the Dolby and dbx systems involve data reduction in the analogue domain. They allow audio signals with a wide dynamic range to be stored on a tape medium with a restricted range — just as we try to squeeze a large amount of digital audio data onto a medium with a restricted data transfer rate. Many of the techniques of analogue noise reduction apply equally to digital audio data reduction, albeit with the far greater sophistication allowed by complex DSP algorithms. Perhaps this is one reason why the masters of analogue noise reduction, Dolby Laboratories, also make some of the most highly regarded digital data reduction systems.

REDUCTION STRATEGIES

Data reduction is used in digital camcorders, DVDs, laser discs, samplers, digital audio workstations, sample-based keyboards, disk-based multitrack recorders, radio and television broadcast networks, telephone systems, and many other applications besides. There is a bewildering variety of systems, many of which have fairly inscrutable names — eg. AC3, DTS, APT X100, MPEG, ATRAC, PASC, and G722. The last of these is an international telecommunications standard for limited-bandwidth speech over the digital telephone network, and is only used in basic ISDN applications. The rest are intended for high-quality stereo (and in some cases surround) audio, and allow the encoding of full-bandwidth signals with wide dynamic range.

To give an idea of the efficiency of these systems, it's worth mentioning that loss-less coding methods (in audio at least) cannot achieve much better than a 2:1 reduction in data rate, whereas the lossy systems can all exceed 4:1 — and some provide as much as 12:1. As a rule, the higher the reduction ratio, the more obvious and detracting are the processing artefacts, but the latest generation of MPEG2 processes running at 12:1 ratios are remarkably good, if not yet completely 'transparent'.

There are four approaches to reducing the amount of data from a linear Pulse Code Modulation (PCM) signal: reduce the sampling rate; reduce the quantising resolution (number of bits); remove redundant data; or remove irrelevant data. The first two approaches offer only modest data reduction if a respectable degree of quality is to be retained, so most advanced systems rely on the last two techniques.

SAMPLING RATE REDUCTION & QUANTISING TECHNIQUES

One of the easiest data reduction techniques is simply to accept a reduced audio bandwidth by allowing a lower sampling rate. Many music samplers, older effects units, and all the digital broadcast systems adopt this approach.

This is a lossy technique, because once the highest frequencies are removed they cannot be replaced. Obviously, the lower the sampling rate, the greater the data reduction, but the price we pay is audio fidelity — the trick is to balance the top-end response against an

Normal Quantising	Non-linear Quantising
111	— 100
110	
101	
100	— 011
011	— 010
010	
001	— 001
000	— 000

Figure 1: By arranging the quantising levels in a non-linear way, low-level signals appear to be quantised to high resolution. In contrast, high-level signals may be encoded with a much lower resolution, but because the signal is loud it will tend to mask the increased quantising noise.

acceptable economy of data. Broadcasters, for example, employ a 15kHz audio bandwidth for their analogue transmission systems, so 32kHz sampling for digital formats such as NICAM and DAB is entirely acceptable, and represents a 40% saving in data rate compared to a 48kHz signal (for more on why the sampling frequency should be at least twice the highest frequency you wish to record/transmit digitally, see part 1 of my series on digital recording in May's *SOS*).

When considering this approach for music recording applications, the sampling rate should be chosen very carefully to ensure that the harmonic structure of the source material is affected as little as possible.

Another simple technique is to reduce the number of quantising levels and thus the number of bits needed to describe each sample, although fewer quantising levels means more quantising noise and a smaller dynamic range. Some early digital effects units employed 8 or 12-bit quantising, for example. This is another lossy technique, as low-level detail in the original audio signal cannot be encoded in a low-resolution system.
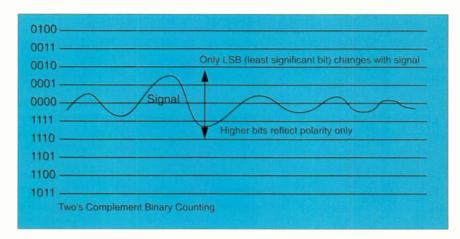
An alternative technique which offers a more acceptable balance between reduced quantising levels and increased quantising noise is to space quantising intervals non-linearly (see Figure 1). The telephone system adopts this approach — it employs 8kHz sampling (giving a 4kHz bandwidth sufficient for intelligible speech) with 8-bit non-linear quantisation. The quantising levels are closely spaced for quiet signals providing a low noise floor (equivalent to a 12-bit system), but as the signal level grows so does the quantising interval. The noise level also increases of course, but since the audio is loud anyway, it will tend to mask the noise.

The technique works surprising well, although noise modulation can become noticeable if the louder audio signals don't contain a wide range of frequencies. A more serious problem is that digital signal processing of non-linear quantised data is almost impossible, because a one-bit increase for a small-amplitude signal represents a very different level change to a one-bit rise in a loud signal.

REDUNDANCY & PREDICTIVE CODING

Another technique is to remove 'redundant data', ie. data that does not carry useful audio information. This may be done in a 'loss-less' fashion — for example, by removing the higher-order bits when a PCM signal is very small (see Figure 2). In this illustration, the 'unused' bits mirror the most significant or 'sign' bit (indicating the positive or negative audio cycle). A quiet signal does not traverse many quantising levels, and the higher-order bits remain static, not conveying anything useful at all — so they can be removed without affecting the quality of the original audio. However, some means of indicating the number of missing bits is necessary so that these can be reinstated before replay. In this way the data rate can be reduced significantly without altering the recreated audio signal at all. An everyday example of this approach is the NICAM television system, which employs this idea as part of its data reduction strategy.

Another loss-less technique is to identify sample values which occur very often in the data stream. These can then be removed and replaced with something to indicate where the true value should be
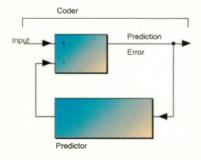
re-inserted. The zero-crossing point, for example, is a very frequent event which can be removed for recording but re-inserted before playback, with significant potential savings.

A more advanced technique, strongly allied to the idea of redundancy, is predictive coding (Figure 3). This is, in principle, a simple idea which has the advantage of suffering very little processing delay (unlike the more sophisticated perceptual coding techniques discussed below), making it a popular technique for real-time applications. Among the real-world systems that employ predictive coding are G722, APT X100, and some versions of the DTS system used in the cinema, laser discs and DVDs.

Audio signals are largely repetitive, which is why predictive coding works. The technique involves a 'predictor' which has a knowledge of typical audio signal behaviour. By looking at the preceding audio signal, it tries to anticipate what will happen next and, because of audio's repetitive nature, the prediction is generally quite accurate. If this prediction is subtracted from the original signal, only a small difference signal remains, and this is recorded or transmitted as the data-reduced result (see Figure 3).

The decoder uses the same predictor 'knowledge' to regenerate the predicted signal. The accuracy of this system is entirely dependent on the predictor algorithm — typically around 98% of the original signal is retrieved.

The technique works less well in anticipating

essentially random signals in noise-like sounds, or in predicting highly unpredictable (but crucial) transients. To improve the precision of the system, therefore, many coders use band-splitting techniques (splitting the whole audio spectrum into four separate frequency bands). This allows multiple predictors to work on simpler band-limited signals with far greater accuracy ▶

Figure 2: With the binary counting system employed here, the higher-order bits mirror the sign bit when a quiet signal is being coded. Thus they are carrying redundant data and can be removed without damaging the audio signal, provided they can be replaced before replay.

HELP!

For more detailed explanations of some of the more complex terms and concepts touched on in this article (such as PCM, quantising, sample rate and bit resolution, to name a few), check out Hugh Robjohns' ongoing series on the basics of Digital Audio, which began in *SOS* May '98 and continues on page 222 of this issue.
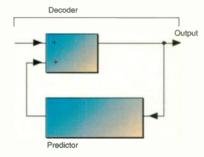
Figure 3: Simplified Predictive Coding structure.

DATA REDUCTION EXPLAINED

▶ than they would if handling the complete signal.

One drawback of predictive coding is that since the decoder must use *exactly* the same predictor as the encoder, improvements to the 'intelligence' of the encoder can only be useful if your decoder is updated too — otherwise the accuracy of the decoded signal will actually suffer.
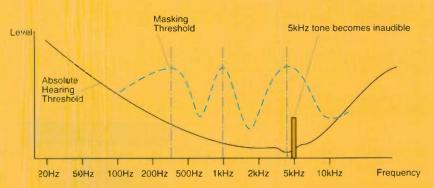
In general, this kind of system works very well, providing a typical reduction ratio of around 4:1. However, it can prove fatiguing to the listener over long periods, because damaged transient signals require more 'brain power' from the listener to interpret the sound. Multiple passes through the encoding/decoding process also lead to rapid loss of signal quality, very similar to that experienced when copying analogue cassettes.

PERCEPTUAL MODELLING

When our auditory systems analyse sound, our brains do not treat the audio spectrum as a continuum. Rather, we perceive sound through around 25 distinct *critical bands* of varying bandwidths (at 100Hz the critical band is about

hiss is inaudible in the presence of full-range music, but obvious between tracks.

The second element of perceptual modelling is temporal or time-masking, in which a loud sound affects our perception of quieter signals both before and after it (see the diagram at the bottom of this box). A quiet signal that occurs 10-20
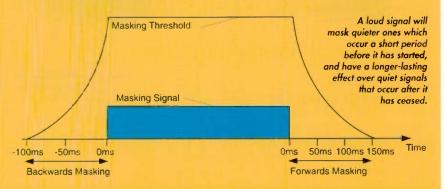
The solid line shows the minimum threshold of hearing against frequency. The dotted line shows how the threshold changes in the presence of three loud tones at 250Hz, 1kHz and 4kHz. Although perfectly audible on its own, a tone at 5kHz (shown as a vertical line) is below the new threshold, and will therefore be masked and inaudible.

160Hz wide, but at 10kHz it is 2500Hz wide). A loud sound within one critical band will tend to mask quieter sounds within the same band, a phenomenon called frequency masking. Although our hearing is incredibly perceptive of simple signals in isolation, in the presence of complex sounds it effectively runs out of 'hearing resources' and so can only perceive the most dominant parts at any particular moment in time (see diagram above).

For example, the hum from a bass guitar amplifier is inaudible while the guitar is playing, although quite evident on its own. Similarly, tape

milliseconds before a louder one, for example, may be masked by the louder signal — this is called backwards masking. The squeak of a kick-drum pedal might be plainly audible on its own, but can be masked by the presence of a much louder bass drum thump which happens a few milliseconds later. The hearing mechanism also takes time to recover from a loud sound, and this creates a masking effect which extends up to 100-200 milliseconds after the masking signal has ceased — this is called forward masking. The length of the masking is related to the amplitude of the masking signal.

A loud signal will mask quieter ones which occur a short period before it has started, and have a longer-lasting effect over quiet signals that occur after it has ceased.

IRRELEVANCY & PERCEPTUAL CODING

The final and most controversial strategy is to remove data which is irrelevant — ie. data representing sounds considered to be inaudible in the presence of the other elements of a complex audio signal. This relies on frequency and temporal masking, and is entirely dependent on the accuracy of a 'perceptual model' of the human hearing system (see the 'Perceptual Modelling' box).

Perceptual coding involves very precise audio filtering and analysis; processing that is only practical using digital techniques. The basic structure of the perceptual coder is shown in Figure 4. A bank of filters divides the audio signal into many narrow bands (typically 32 or more) prior to processing, and a perceptual model then analyses the spectral content of the audio to determine which elements are likely to be completely masked (ie. irrelevant) and can therefore be discarded. The remaining audible signals are re-quantised with a low resolution just sufficient to put the quantising noise below the masking threshold in each frequency band.

Temporal masking is calculated by dividing the signal up into blocks of samples (typically around 10 milliseconds in length) and analysing each block for transients which will act as temporal maskers. Most systems vary the length of the block to take advantage of both backwards and forwards masking (explained in the 'Perceptual Modelling' box elsewhere in this article).

Perceptual coding's complex digital filtering and spectral analysis, combined with the process of treating the audio in blocks, can create significant time delays in the audio signal. A complete encode/decode path can exhibit anything between 20 and 200 milliseconds of delay, which often causes serious problems in real-time applications.

The decoder is much simpler than the encoder, as it does not require any perceptual modelling knowledge at all. It simply has to decode incoming data back into the original filter bands and re-quantise the data to conform with a standard PCM format (based on codes embedded in the data), before re-assembling the data into a composite output signal. Upgrades to the perceptual model in the encoder will automatically result in better quality from every decoder, purely through refinements to the process of deciding which bits of the signal are irrelevant.

MPEG1 and 2, PASC (used on Philips' Digital Compact Cassette), ATRAC (used on Sony's MiniDisc), and Dolby's AC3 are all systems which use perceptual coding as their primary means of determining irrelevancy in complex audio signals. However, they can all work at a variety of sample rates, they all employ requantisation to reduce the number of bits per sample, and they all remove some degree of redundancy in the signal too (see the 'Perceptual Coding Processes' box for more on these).

GOOD OR EVIL?

A great deal has been written and said about data reduction systems, much of which has suggested that they are an unnecessary evil. This is not the case: provided the data reduction system is appropriate to the application and is used sensibly, it is a useful ▶

DATA REDUCTION EXPLAINED

► tool that allows better results than would otherwise be possible — for example, longer recording times, smaller recording media, more tracks, or whatever.

This is not to say data reduction is suited to every situation, but digital recording with most modern data reduction systems will provide better quality than most semi-pro analogue recorders, especially cassette multitrackers.

Multitracking is a specific case where digital data reduction offers profound advantages. Since each track is carrying relatively simple signals which include a great deal of irrelevancy and redundancy, data reduction

systems can work very effectively in allowing more tracks to be squeezed onto a particular medium with negligible side effects. Data reduction systems usually only reveal their weaknesses with very complex signals, so you might prefer to record your final mix on a linear PCM format, but data reduction can offer more benefits than drawbacks for the multitrack recorder.

One of the biggest problems with current data reduction systems is in coding stereo (or surround) material. This is primarily because many perceptual models are not sophisticated enough to cope with 'stereo unmasking'. This is a phenomenon whereby, although a quiet signal in one channel might be masked by louder elements in the same channel, in the context of stereo monitoring its spatial position (determined by the relative levels between the two channels) effectively unmasks it, revealing it as a separate and identifiable signal.

Imagine, for example, a rhythm guitar panned three-quarters towards the left, behind the rest of the instrumentation in a complex mix. The perceptual coder might decide that the (quiet) guitar in the right channel is masked by everything else going on, although it is sufficiently loud in the left to remain audible. Consequently, the guitar might be removed from the right channel as part of the data reduction with the result that, when listening in stereo, it will appear to be positioned hard left instead of three-quarters left. In practice, it is more likely that some frequency components of the guitar (for example harmonics) will be removed while others are retained, and so the image of the guitar will become blurred. Although this is perhaps an exaggerated example, I hope the point is clear: data reduction systems can impose instability in stereo images if they are not very carefully designed.

One solution is to treat the signal as M-S (middle and side) components rather than L-R (left-right) components. This takes advantage of the fact that most loud signals sit in the centre of the stereo image (middle in M-S) and are therefore common to both channels, leading to a lot of redundancy. By dealing with stereo in this way, a greater proportion of the data rate can be allocated to the side signals that convey imaging information. This is often called Joint Stereo Mode.

THE BOTTOM LINE

Data reduction systems necessarily reduce the amount of data, so it stands to reason that multi-generation processing will lead to quality loss. The MPEG algorithms seem to be slightly less prone to this, but it remains a general problem with all systems. Things are worst when different data reduction systems are cascaded, in which case the quality deteriorates very quickly indeed. The symptoms usually become apparent after two or three generations in the form of vague stereo imaging, noise modulation, a harshness to the sound, aliasing, and increased background (quantising) noise.

In general, it is wise to avoid recording a complex mix on a data-reduced format if any subsequent processing or copying is envisaged. It would also be sensible to avoid recording stereo source material on data-reduced systems wherever possible, but single mono sources (eg. individual tracks on a multitrack) will not be noticeably affected by data reduction. ■SOS

PERCEPTUAL CODING PROCESSES

• MPEG1 & MPEG2

MPEG is an abbreviation of 'Motion Picture Expert Group' — an international body set up in 1988 to define digital audio and video data reduction systems. MPEG1 is an elaborate perceptual coding specification with three subdivisions of increasing complexity. Layer I is the simplest, offering typically a 4:1 reduction in data rate with a 32-band filter bank. Layer II is essentially the same thing, but employs more complex spectral analysis of the input signal. This allows more accurate perceptual modelling and thus greater data reduction. Layer III goes further and incorporates even more sophisticated techniques such as varying filter bank bandwidths to better simulate the critical bands in human hearing, and non-linear quantising to increase the efficiency of the data reduction.

MPEG2 is an extension of MPEG1 providing multi-channel surround sound capabilities such as 5.1 channels, although other arrangements are also supported. The original MPEG2 was designed to be fully backwards compatible with MPEG1 systems, although a non-backwards compatible version (MPEG-2 NBC) has recently gained approval for DVDs and broadcast applications.

• PASC

Precision Adaptive Sub-band Coding is derived from the MPEG1 Layer I audio data reduction system. It is used in the Philips Digital Compact Cassette (DCC) and operates at a compression ratio of 4:1.

• ATRAC

Adaptive TRansform Acoustic Coding is the system employed by Sony on their Minidisc as well as in the SDDS cinema surround sound format. It offers a 5:1 data reduction ratio in the case of Minidisc and employs the equivalent of 52 filter bands for spectral analysis and re-quantisation. The size of the sample blocks is varied dynamically between 11.6 and 1.45 milliseconds according to the nature of the audio signal to accommodate temporal masking. ATRAC has now gone through many revisions since its launch, and the latest versions are extremely good.

• AC3

This is Dolby's third data reduction process (hence the name) and is already widely employed in laser discs, DVDs, digital television broadcasts, and SR-D/DSD cinema presentations. It differs from the other systems mentioned here in that it was designed from the outset to accommodate multi-channel audio formats with data rates ranging from 32 to 640 kilobits per second per channel depending on the application.

I f you've been involved in recording (and reading *SOS* regularly) for the past few years, you won't have failed to notice that for some time now, desktop computers have been capable of taking over the recording functions that were traditionally associated with dedicated studio hardware. For example (so the theory goes, anyway), many current PCI computer soundcards have everything required to turn your computer into an all-digital 'recording studio in a box'. Add a latest-generation MIDI + Audio recording/editing software package, and you might begin to wonder why you would need any 'traditional' hardware in your studio.

In reality, though, there is a world of difference between what you can achieve with the 'everything in a single computer' approach and what is expected of a professional studio. Of course, you do find desktop computers in top-notch studios, but they don't use 'multimedia' soundcards — and they aren't expected to do everything.

So why can't you necessarily expect top-end performance from a soundcard-based computer studio? After all, if it's all digital, shouldn't the sound be great regardless of the platform? Well, one of the problems is that 'real' sound sources (ie. acoustic ones, such as guitars and people's voices, as opposed to electric or electronic ones), are *not* digital. They have to be converted from analogue sound to digital data, and that is actually something that a conventional PC soundcard typically does rather badly.

Let's say you have a 16-bit soundcard in your PC. What this means is that a signal that stops just short of overloading the input to the card will result in a digital output peaking at 16-bit resolution. Lower-level signals are encoded with fewer bits and it is often the case that the bits representing the lowest signal levels do nothing at all, but just flip randomly, creating noise. Added to this is the cacophony picked up by the analogue part of the soundcard from the computer itself (in audio terms, the inside of a computer is just about the worst possible environment, because almost everything that happens in there generates a spike or a square wave). Listen critically to the audio outputs of almost any soundcard with onboard converters and you will be shocked at the level of computer-generated noise.

What's the answer? Well, one idea is to locate all vestiges of the analogue world outside the computer: and that includes the analogue-to-digital and digital-to-analogue converters. I've been arguing for a long time now that as the audio business converges with the computer industry, the focus of audio skills will need to be directed at programming, and designing high-quality input/output devices. This prophecy seems slowly to be coming true, and the new Aardvark Aark 20/20, which styles itself as a Professional 20-Bit Multitrack Hard Disk Recording System, is clear evidence of this.

WHAT YOU GET

Aardvark is a small American pro audio company based in Michigan, USA. They already have a healthy reputation as producers of the AardSync II low-jitter master digital audio clock generator for high-end digital audio workstations (reviewed in *SOS* October '97) and some low-jitter clock distribution equipment. ▶

Multichannel PCI-card based digital audio recording systems are increasingly common these days, but not all of them focus on the quality of the onboard converters to the extent they should. DAVE SHAPTON checks out a new card from far-sighted digital problem-solvers Aardvark that promises not to throw out the baby with the bathwater.

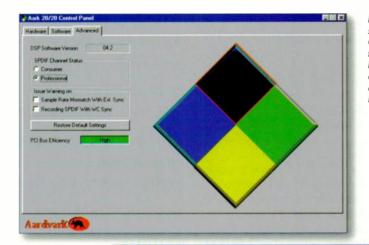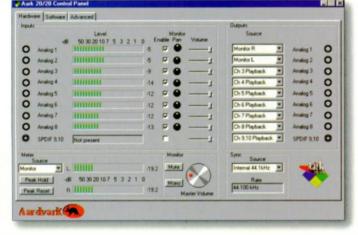# Heavenly Vision?

AARDVARK AARK 20/20 20-BIT PC DIGITAL AUDIO RECORDING SYSTEM

Aardvark Aark 20/20

pros & cons

AARDVARK AARK 20/20 £999

pros
- Very solidly built.
- Transparent sound.
- Built-in I/O routing matrix.

cons
- Doesn't offer some of the software 'bells and whistles' of its multichannel competitors.

summary
With its high-quality 20-bit converters and simple-to-use but fully-featured control panel software, the Aark 20/20 is ideal for use as the basis of a digital recording system. All it really lacks at present are native drivers for specific audio recording applications, but even these are on their way.

SOUND ON SOUND

▶ The Aark 20/20 system consists of a very solid, nicely made PCI card with an onboard DSP chip (the so-called Host card) and an external box containing eight channels of both 20-bit A-D and D-A conversion. S/PDIF in and out is also provided, and can be used at the same time as the eight channels of analogue I/O, giving a total of 10 channels that can be used simultaneously. Software drivers for Windows 95 allow audio applications to 'see' the Aark 20/20 as if it was five stereo soundcards, so most modern software should work just fine with it. To get you started, the card is bundled with *Samplitude Studio* audio recording package from the German Company SEK'D.

INSTALLATION

I have to say that this is one of the easiest audio products I have ever installed. Having a virgin Windows installation with no other exotic PCI cards (apart from a generic S3-based video card) probably helped, but I have had enough nightmares with other audio devices to make me grateful for a nice, clean, simple installation. In case you have never installed a product like this, the documentation supplied is helpful, with several screenshots showing how it is done.

I tested the installation routine on both Windows 95 and Windows 98 and found both to work equally well. There are no Windows NT 4.0 drivers yet, but they are being worked on, and ASIO drivers are close to release, which will boost the product's responsiveness with *Cubase VST*.

The supplied Aardvark software control panel (from which the screen dumps that adorn this article are taken — see Figure 1) tests your PC's suitability for multitrack audio work by testing the efficiency of the PCI buss. The PCI capacity of my generic Pentium II 266MHz was rated as 'High'. This means my system should be able to simultaneously record and play back 10 channels of digital audio, assuming I have a fast enough hard drive. Once the Host card is installed, a single 2-metre shielded 25-pin cable connects the PCI card to the back of the Aark 20/20 converter box. This cable also provides power to the external unit. Most PCs are populated with at least one other 25-pin socket, so some care is needed to find the right connection (I'm rather sensitive about this sort of thing, because I once blew up a motherboard by confusing a SCSI port with a printer socket...)

The sturdy external unit has two rows of eight quarter-inch connectors on the front, with outputs on top and inputs below. Beside these are phono sockets for S/PDIF in and out. On the back, along with the host PCI card cable, are BNC connections for word clock in and out, together with a BNC connection for video in, ready for a future upgrade to enable the Aark 20/20 to derive its master sampling rate from an external video signal. The word clock out can be used to synchronise the Aark 20/20 with other digital audio devices, including a second Aark 20/20. Indeed, the software automatically recognises the presence of a second unit and configures itself for 16 tracks of analogue I/O instead of eight.

DSP & ROUTING

The system comes complete with onboard digital signal processing; but the Aark 20/20 makes no claim to be a digital effects device. The DSP is actually used to increase the flexibility of the 20/20 in some very useful ways. Firstly, it can behave as a digital patchbay; any input can be routed to any or every output via the software control panel. In the digital domain, it's easy to route a single input to multiple outputs; but if you want several sources to be routed to a single destination, then you need digital signal processing. Although the Aark 20/20's main role in life is to get eight discrete analogue outputs to an external mixer, the ability to create submixes and a separate monitor mix within the unit is incredibly useful. The monitor mix is a separate function and can itself become an input source for a piece of recording software. This means that you can use it to mix all eight inputs (or all 10 if you use the S/PDIF inputs as well) into stereo and then place them on just one pair of tracks in your software recording application. The level of each track can be controlled individually, and there is a master mix volume control as well, and a pan control is planned for the next version of the software (see Figure 2). You could also use the application itself as the monitor signal source and have the Aark

20/20 bounce up to 10 tracks to another pair of tracks. The monitor function is best seen as an "eavesdropping" device that stops short of being a fully functional mixer. It is, though, an extremely handy utility.

Another use for the DSP on the Aark 20/20 is to generate a test tone. All outputs can select 'Tone' as an option: very useful when you are setting the system up and just want to check signal paths and levels. 'Silence' is another DSP option that has less obvious uses, but can be used to demonstrate the noise floor of the system if it is selected as a signal source. The software control panel supplied with the Aark 20/20 is visually unexciting, but this is hardly a problem when you consider that its main function is to set up the routing of the inputs, outputs and monitor buss. Three tabs at the top of the control panel let you configure the hardware, software and advanced settings such as professional or consumer S/PDIF (some DAT machines are very fussy about the somewhat superfluous tag in the S/PDIF bitstream that distinguishes between the two formats). The software tab reveals what I mentioned earlier; that the Aark 20/20 appears to a recording application as five stereo soundcards. This is becoming common practice with multitrack soundcards and is a good way to gain compatibility with the largest number of applications. I noted that when the Aark 20/20's control panel was maximised, it consumed a significant portion of my PC processor's number-crunching capability. It looks to me as if this is due to the real-time meter display. It's not a problem as long as you don't maximise the control panel while you are playing back a lot of tracks.

NEVER MIND THE BITS, FEEL THE QUALITY

The big advantage the Aark 20/20 has over other PC multitrack I/O units is that the converters are outside the PC. This should lead us to expect a very low noise floor, since all of the sensitive analogue parts are well away from the PC and shielded from it. When I test converters I always make a point of listening closely to the amount and type of noise output when there is no signal present. On a typical PC soundcard with internal converters, you don't have to raise the monitor levels very far before you begin to hear a din that resembles an electronic dawn chorus. Not so with the Aark 20/20. At all normal listening levels, the output seems completely silent. At the very highest gain settings, some noise can be heard, but even then it is just that — noise, with some, but very little, of the 'graininess' that seems almost inevitable with digital converters. Overall, then, this is a very quiet unit.

As far as the audio quality is concerned,

I have to say that these converters have very little sound of their own; but then that's exactly what you want from converters! Sure, I wasn't listening in a perfect environment, but my setup is probably typical of a digital project studio: £400 studio nearfield monitors, good speaker stands and good cables, in a well damped room. The gain through my review 20/20 was a few dB less than unity, so it was difficult to make a direct comparison by switching the unit in and out, the problem being that the difference in volume completely masked any qualitative difference there may have been. (I didn't want to introduce any additional gain stages to compensate, because that would have affected the sound as well). With practice, I found I could make the switch and adjust the volume all in one action, thereby giving me a valid comparison between direct audio and audio through the Aark 20/20.

Essentially, then, these converters are very nearly transparent. If my life depended on my being able to describe the effect they had on the sound then, yes, the bottom end lost a little clarity. But to put this in perspective — I had to cascade all eight inputs and outputs to confirm that this was the case. Even with the audio undergoing four conversions from A-D and D-A (I wanted to keep the signal stereo) the result was still better than some converters manage after just one conversion! At all times, the sound was remarkably smooth; lacking any sort of harshness, yet never short on detail.

THE FINAL WORD

Aardvark is a company that makes products to solve digital audio problems. It is clear from the design and quality of the Aark 20/20 that they know about digital audio. Apparently, hidden under a sealed cover on the Host card is a technology unique to Aardvark that uses digital signal processing to reduce jitter, which is one of the most insidious forms of distortion in digital audio. It is this kind of attention to detail that you should take into account when choosing a multichannel converter. You shouldn't base your decision just on the number of inputs and outputs. After all, you don't choose a car because it's got four wheels, do you? SOS

An eight-buss console with pro looks, serious spec, and even a couple of valve stages too, all at a project studio price; surely it sounds too good to be true? Not when it comes from Allen & Heath, as DAVID MELLOR finds out.
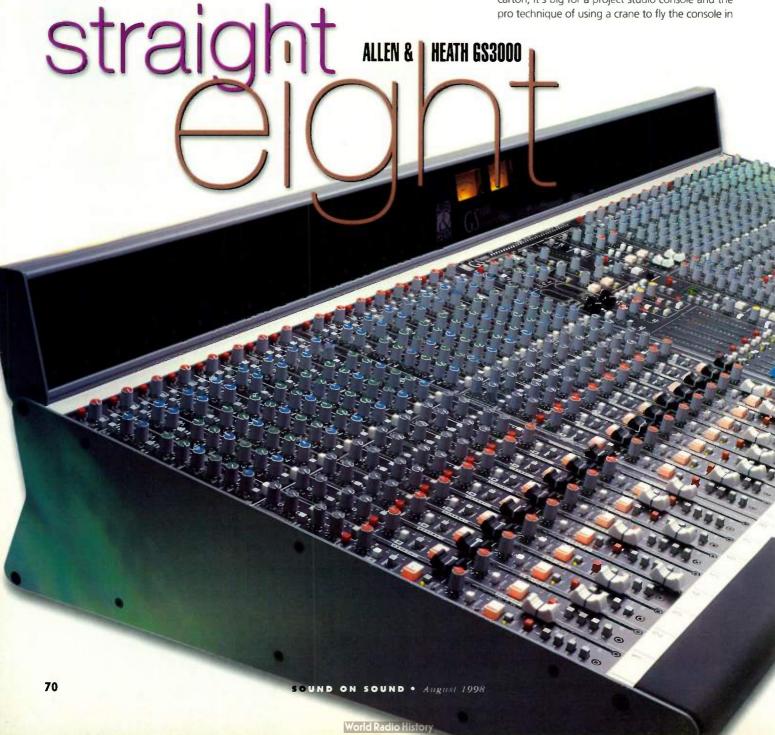
Every home and project studio owner aspires to have a 'proper' professional mixing console as the centrepiece of their studio — something like an SSL J Series would do nicely for me. Most of us, however, can't afford that kind of price tag (although Paul Allen, who founded Microsoft with Bill Gates, has a top-dollar AMS-Neve Capricorn digital console in his home studio, just for fun you understand). We tend to have to make do with something smaller, cheaper, less versatile, and too often somehow lacking the 'aura' of a real pro console. But help is at hand... Allen & Heath evidently decided that they would seek to design a console that would fill precisely this gap, and at a reasonable project studio price too. The result is the GS3000. Quite simply, it looks the business and anyone with any experience in the world of pro studios could walk into a GS3000-equipped studio and feel very comfortable because they can see at a glance that the place is properly equipped. And as we shall see, the GS3000 doesn't just look the business, it *is* the business. It's a console for the professional,

not the dabbler, and virtually every pro requirement is catered for, with lots of additional neat touches that demonstrate that the GS3000 is capable of turning out commercial product, day in day out.

In an era when digital mixing consoles are becoming increasingly powerful and affordable, the Allen & Heath GS3000 is absolutely rock-solid analogue. See this as a disadvantage if you like, but you can at least be assured that this console is 24-bit/96kHz ready, which is more than any of the current crop of digital consoles can claim! Traditional analogue design also has the advantage that any engineer can walk into your studio and start recording or mixing. No training, no manual, just roll the tape and push up the faders. Although traditional analogue consoles will always lack the programmability of digital consoles, familiar appearance and methods of operation are very valuable features.

IT'S BIG

Be warned that the GS3000 comes in a very big carton; it's big for a project studio console and the pro technique of using a crane to fly the console in

straight
ALLEN & HEATH GS3000
eight

through a window may — almost — be appropriate in cramped locations (I didn't have enough headroom in my studio to stand the empty carton on end!). The GS3000 is itself a very satisfying size, although with respect to the wide range of facilities it provides, you could call it compact. The 32-channel version is just under a metre and a half wide and since it weighs 37kg you'll need someone to help get it in position. The construction of the console doesn't betray any of the obvious cost-cutting measures that are commonly employed on project studio consoles. The connectors are exactly where they should be — on the rear panel — so the cables are not visible for all to see, nor are they only partially hidden by the bolt-on meter bridge (the meter bridge on the GS3000 is actually an option, but once in place it looks like an integral part of the unit). The pots are all bolted to the front panel so you can be sure that no stress is being placed on the printed circuit board behind. Internally, the construction is modular, so that servicing is easier — a circuit module could be sent for repair while the rest of the console was still being used as normal. Another nice touch is the arm rest at the front. Unfortunately I can't tell you that that it is padded leather, but it is at least slightly soft and provides a useful degree of creature comfort for protracted mixing sessions. The power supply is external, as is appropriate for mixing consoles, and is silent! Great big heatsinks on the back perform the work that is too often taken care of by a fan.

The Allen & Heath GS3000 is an 8-buss console — a very popular configuration since the combination of eight mixable outputs and direct outputs from the channels (which the GS3000 offers) is enough to cover virtually every situation you are likely to come

"Anyone with any experience in the world of pro studios could walk into a GS3000-equipped studio and feel very comfortable."

across in music studio recording. 24-channel and 32-channel frame sizes are offered which give, respectively, 52 and 68 inputs to the mix. The topology is in-line, meaning that each channel strip contains the monitor controls for one channel of the multitrack, as opposed to the (now quite rare) 'split' monitoring design where the multitrack monitors are somewhere else on the control surface. As a surprising bonus, the GS3000 includes a two-channel valve stage which can be patched into the console wherever necessary, and can also be used as a guitar input preamplifier. I have been saying for a long time that mixing consoles should have high-impedance inputs (as well as the traditional mic and line inputs), suitable for electric guitars and pianos etc. Here, at last, we have them and with valve warmth too! Although the GS3000 doesn't have fader automation (which you could hardly expect in its price range) it does have the potential for mute automation, which is really half the battle when it comes to mixing band recordings, and in addition it offers integrated machine control. This means that your multitrack(s) can be away in a machine room somewhere while you issue all the necessary transport and track-arming commands right from where you are sitting. Integrating machine control and automation into the console was one of the features that helped Solid State Logic to make their big breakthrough in the top flight — Allen & Heath have made a similar breakthrough at the project studio level with the GS3000. Of course, machine control is nothing if it doesn't give you the functions you need and isn't easy to use, and we'll see later whether it does or not. But first...

THE CHANNEL

Reassuringly normal. That's what many users, and virtually all analogue enthusiasts will want me to say. And it is. The GS3000 displays classic design and incorporates the best thinking of the last 30 years to provide a channel strip which is as straightforward yet effective as it possibly could be in its price range. Let me point out some of the interesting features, starting from the top. I was pleasurably surprised to see the gain of the line input ranging from -10dB to 40dB. The negative figure is actually of most interest here because it helps when something is coming in to the console at a high level and yet needs to be mixed at a low, but precise level. Being able to set the gain to a negative value allows the fader to be higher up in its travel where it has better resolution. Unfortunately this flexibility doesn't extend to the tape input which is merely switchable between +4dBu/-10dBV. The insert point is pre-EQ. On a console such as this, I believe it should be post-EQ (see 'Channel Inserts' box), but I can live with it.

The GS3000's EQ is, apparently, of the 'British' variety. Fair enough, Cornwall has been a part of Britain for as long as I can remember, so I suppose it has to be true, but this term, 'British EQ' really goes back a long way to the days when some Americans thought that certain British consoles had better EQ than their home-grown product. These days, every competent console manufacturer knows everything there is about equalisation in a technical sense and it is now an economic and functional ▶

IT'S HOT

Valve mixing consoles in current production are very few in number, and the GS3000 doesn't claim to be one, but it does have the benefit of a patchable valve 'warming' stage. Being patchable, the two channels of valves don't affect the sound quality of the GS3000 unless you want them to, and there are two ways of using them. One option is to use them as inputs where they will work as mic or line inputs (with phantom power for microphones). The valves are driven in a symmetrical mode where the positive and negative half-cycles of the waveform are distorted equally giving a pleasant warming effect. Alternatively, they can be switched to guitar mode where a guitar may be directly connected and only the positive half-cycle is clipped, leading to more obvious distortion. As well as using the valve stages as inputs to the console, they can just as easily be patched to insert points in the channels, groups or masters. So you can warm up a single channel, or the entire mix, if you wish. It's an interesting feature — not essential perhaps, but I bet you will use it.

ALLEN & HEATH GS3000

decision as to what facilities, frequencies and slopes are provided in a console design. For many people, mixers still stand or fall by their EQ, but Allen & Heath's GS3000 can certainly stand tall, at least in the project studio sector. Two mid-frequency sections are offered with *full* parametric control. This means that each band has controls for frequency, gain and Q — and it's a rotary control for Q, not a switch. Q ranges from 0.6 to 2, which still isn't quite as low on the low side as I would like (0.3 please), but apart from that, this is a good feature. The Q setting actually changes the range of frequencies over which the mid-band gain control is effective — a low Q means a wide range of frequencies. High values of Q are most appropriate when drastic action of some sort is called for, either correctively or creatively. On this console, a Q of 1 is in the centre position of the control and it is detented, which I approve of. The other EQ facilities include basic HF and LF controls. Here, unfortunately, you don't get a choice of switched frequencies. This is a shame, but you can't have everything and a reasonable price tag as well. There is also a 100Hz low-cut filter and the all-important EQ Out switch, for reducing circuit noise (by a very slight, but useful amount) when EQ isn't needed, and for making comparisons with the flat signal.

Moving down the strip, the auxiliary sends look rather less capable at first sight than they actually are. There are four knobs, but the lower two can be switched (on an individual channel basis) to address aux outputs 5 and 6. Auxes 1 and 2 can be globally switched to either pre-fade or post-fade. Global switching of auxes is, of course, a cost-cutting measure, but it is, I think, a reasonable compromise

The GS3000's patchable valve stages can be used as mic or line inputs, or as a high-impedance guitar input.

here and any problem you might have can always be worked around by some other means. Auxes 3 to 6 are all post fade, which is most appropriate for reverb and delay effects. A red button close to the aux sends curiously labelled 'XFX' provides an additional stereo aux send when mixing.

The lower section of the channel strip is actually the most interesting and exciting part. This is where Allen & Heath have decided that the GS3000 will mimic a high-cost commercial studio console as closely as possible. Top pro consoles, as you will almost certainly have noticed, have two faders per channel strip. Project studio consoles just have one. Well the GS3000 has the full complement of two, and is all the better for it. The small upper fader controls the level going to the multitrack recorder. The large lower fader controls the monitor level of each track during recording and overdubbing, and the level of each track during mixdown. Yes, what I said is correct — the small fader is the multitrack send and the large fader is the monitor. This is the ▶

The in-line channel strip is reassuringly conventional.

CHANNEL INSERTS

If the channel insert point comes before the EQ then it is easy to set up a noise gate. Once you have found the correct threshold level, it will stay exactly the same and gating will be reliable, whatever else you do to the mix (apart from adjust the gain control, which you wouldn't have to do if you set it correctly in the first place). But many engineers find it better to insert a compressor after the EQ — subjectively, it often sounds better, and there are good reasons why it should. Allen & Heath have chosen the first option and placed the insert point before the EQ. But the thing is that the muting system, with the potential for mute automation, is so good that you would hardly ever need need to use a noise gate. The insert point could have been post-EQ and we would then have had the best of both worlds.

MUTE AUTOMATION

Mute automation is great when you are recording a band as lots of unwanted noises tend to get onto the multitrack before, during and after the song. With MIDI instruments, mute automation isn't quite so important, although it can still be useful to create 'breakdown' mixes to edit into the finished song later, perhaps for an extended remix version. Although the GS3000 itself provides mute grouping and mute patching, an external MIDI sequencer is necessary to automate the muting. Mute grouping is handled by four buttons in the centre panel. Combinations of mutes can be set up and stored under each of the four buttons so that you can turn on or off any number of mutes with a single button press. If you have never used mute grouping you won't know how wonderful a feature it is — believe me, it's very wonderful! Four mute groups is a good number, but there are times when you will want more. This is where mute patching comes in. It's a very slightly longer procedure to set up and recall a mute patch, but you can have up to 128 different patches which should be enough to cover any situation. These patches correspond to MIDI programs which can be recorded into and played back from a MIDI sequencer.

ALLEN & HEATH GS3000

The GS3000 offers a comprehensive connector array.

▶ exact opposite of the traditional in-line console, but it is actually becoming increasingly common, and when you think about it, it does make a lot of sense. You can build your mix as you go along, and there is no need to switch from the small to the large faders when you start mixing since you are there already! For some curious reason, although the large fader is smooth and easy to operate, the small fader is quite stiff. They do their job though. The group routing buttons are, a little confusingly, located next to the monitor fader (I still hate this clutter around the faders. I always have and I probably always will). They can, however, be switched to either the channel or the monitor signal path so perhaps this is reasonable. Both channel and monitor faders have large LED-equipped mute buttons controlling and signalling whether the source is sending signal to the busses. Big buttons with LEDs are indeed pro features because you need to see what's active and what's not, at a glance. These buttons don't latch because they can also be controlled by the muting system, which is a very powerful facility when mixing recordings of bands and other non-MIDI combinations of instruments. The console also offers Solo-In-Place as well as conventional PFL. There is also an alternative Mix B buss with a number of possible uses for the creative engineer. Both of these are important features.

CENTRE PANEL

Oddly enough, there is no such thing as an auxiliary return on this console. At least there is nothing called an auxiliary return but there does just happen to be four stereo input sections. I like this approach because in my alter ego (one of several) as an audio educator I find it difficult to separate auxiliary sends and auxiliary returns in people's minds and there is a tendency to think that it is obligatory to patch the output of a reverb unit into an auxiliary

return when, if the project calls for it, it can be just as easily connected to a couple of spare channels thereby enabling it to access the full range of EQ and other features. It may be just a conceptual matter, but I think Allen & Heath's presentation is correct. In fact, two of the stereo inputs are almost as fully featured as the mono channels, with gain, 4-band EQ (but with fixed mid frequencies) and all six auxiliary sends available. Full-size faders are the icing on the cake. The other two stereo inputs are more modest in capability, with only two of the aux sends (auxes 1 and 2) and small faders, but I can certainly find no fault here.

To complete all of the console's inputs before moving on, the GS3000 provides 2-track return inputs for three stereo recorders. This means you can have a DAT mastering machine, a cassette for 'quick and dirty' copies and a CD player for reference purposes, all permanently patched to the console ready for use at the push of a button. More than this, the console provides push-button routing so that direct copies can be made from Stereo 2 to Stereo 1 and vice versa, and from Stereo 1 to Stereo 3. It's a small thing, and Allen & Heath are not going to sell any more consoles because of it, but it shows they are thinking of real-life requirements and what really goes on in a session besides the fun stuff.

Having dealt with the inputs, let's look at the GS3000's outputs. There are eight groups, as you would expect from an 8-buss console, and they can be switched to work as subgroups feeding into the stereo mix buss if required. The group faders are calibrated, like the channel faders, up to +10dB which means there is always some extra gain in hand in case the meters on the multitrack are not quite hitting the end stops! Actually my personal preference is for group faders that have 0dB as their upper limit, as the master fader does here.

It is common practice these days in mixing console design to route the group outputs internally to the channel strips, in banks of eight, so that Group 1 feeds tape send outputs 1, 9, 17 and 25. I would have liked a set of eight outputs actually labelled Group Outputs since it would have been practical to wire these to a patchbay and wire the tape outputs directly to the multitrack to give the optimum compromise between flexibility and economy of wiring. You could still wire tape outputs 1-8 via a patchbay, but they would be affected by the Group/Direct switch on the channel which could cause confusion. Insert points are provided on the groups and on the masters. Auxiliary outputs, as you would expect on a console of this nature, do not have insert points so if you want to compress the send to the reverb from a mix of channels (what do

you mean you've never tried it?) you will have to find another way of doing it. Typical, isn't it? Allen & Heath have gone out of their way to provide such a wonderful range of facilities and I'm *still* asking for more!

The GS3000 is very well provided for in the monitoring department. Once again, Allen & Heath have covered almost every scenario. Any console will provide control-room monitoring, with outputs for a power amplifier and headphones. Many consoles will provide alternate-speaker switching so that main monitors or nearfields can be selected. Many consoles will also provide a mono switch so the mono compatibility of a mix can be tested. Fewer consoles will provide all of the aforementioned and, in addition, studio monitoring so that you can play back a take over speakers in the studio and save the band a trip back to the control room to hear their work. The GS3000 goes one stage further and provides *two* studio monitor outputs, so if your studio has a drum booth annexed to it the drummer can have his own loudspeaker playback too. In fact, rather than providing loudspeaker playback in the studio, which is commonly done at a professional level, the way you will probably use the studio outputs is for foldback on headphones. The classic way of doing this is to use a pre-fade auxiliary send, but this takes time to set up and many people take the quick route of patching the main stereo mix to the headphones. The studio outputs of the GS3000 can be driven by a number of sources: auxiliary 1, the main stereo mix, Mix B or the control room monitor. The great thing is that you can mix these sources together, so you can quickly select the main mix as the monitor source, then if the singer (for instance) wants to hear more of a particular instrument you can add it using aux 1. If you really want to be clever you can invert the phase to 'add' less of a particular channel. Talkback, mute and AFL are also provided.

A PROFESSIONAL SOLUTION?

The sound quality of the Allen & Heath GS3000 is fully in line with contemporary mixing console design for this sector of the market and I am sure you will be entirely satisfied. The muting and machine control facilities work well and you will probably wonder how you managed without them. The EQ is effective and I am sure most users will find it musical. The incorporation of a pair of valve stages may be considered something of a novelty, particularly since there is plenty of outboard equipment available that can do the same job, but it is certainly perfectly usable, and having the option of high-impedance inputs on a mixing

console can only be a good thing. Analogue it may be, but the sheer range of facilities, and the way these facilities have been tightly directed at the serious professional user, make the Allen & Heath GS3000 a major contender in the project studio, and even the smaller commercial studio market. **SOS**
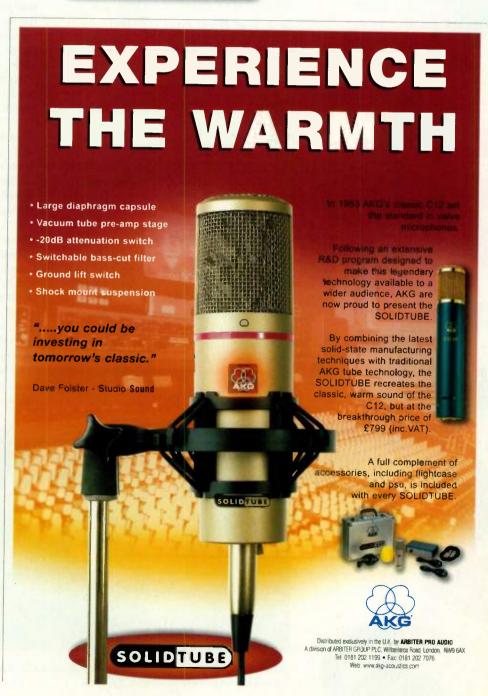
Recording Guitars

*PAUL WHITE and
DAVE LOCKWOOD strum
up a few tried-and-trusted
methods of improving your
studio guitar sound.*

At one time, recording electric guitar meant putting a mic in front of an amp and hitting the record button. Of course, you can still do it that way, and in many instances such an approach yields perfectly acceptable results. But there are also viable DI alternatives that may be more convenient in a home recording environment.

The fact that guitar sounds vary so much according to the instrument, the player, the amplifier and the recording techniques employed is one of the great attractions of the instrument. But while no two players will ever sound quite alike, there are a few general recording rules that can help achieve consistently good results. It would be wrong to try to define a single 'foolproof' method for guitar recording, so instead we've pulled together a few different tricks, tips and ideas that will help you get the results you want, while still providing plenty of scope for experimentation.

As ever, if you get a result by breaking a rule or doing something in an unorthodox way, that's fine. In music, the end always justifies the means!

• As a first and very basic rule, always make sure your guitar is in good condition. Use new strings, properly fitted and check both the tuning and intonation. This may seem obvious, but many players leave their old strings on, tune up and hope for the best. They think the tone doesn't matter because you'll be able to fix it in the mix. More often than not you can't! When fitting new strings, make certain that you don't have any overlapping turns of the string around the tuning machine peg as the tuning will tend to slip. Also, pull the strings to stretch them before final tuning or again, the tuning will slip. If you change either the brand or gauge of strings you're using, check the octave tuning on all six strings to ensure the intonation is OK.

• Check your tuning before every take, as it tends to drift in warm studios. Ideally, use a tuner with an audio thru socket and leave it connected at all times. Users of vibrato arms should be especially careful to check their tuning at every opportunity.

• Equipment containing transformers will tend to cause hum interference on electric guitars, especially those with single-coil pickups. Rotate your position in the room to find the angle of least hum and keep as far away as possible from the interfering equipment. As a rule, the more overdrive you use, the more serious the effect of interference. Computer monitors in particular cause serious interference problems with guitar pickups, so keep as far away from these as you can. Or better still, turn the monitor off. (Note, though, that if a monitor is in energy-saving mode and is still powered up, the screen may be dark, but you'll still get interference.)
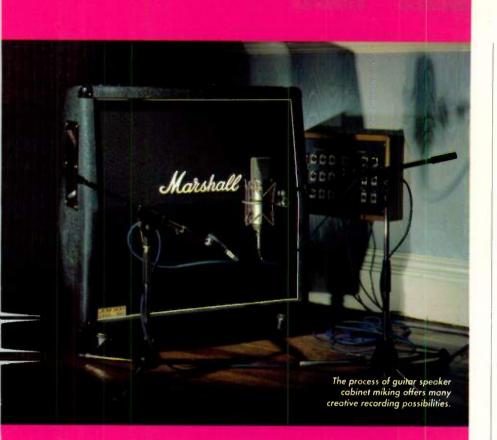
• Use different guitar types or sounds when recording two or more overdrive guitar parts to keep the sounds separate in the mix. For example, use a single-coil setting for one part and a humbucker for another. It also helps if you use less distortion than if you were playing live, especially for rhythm

The Palmer Junction passive speaker simulator/DI will accept a speaker-level input, but does not include a dummy load.

The process of guitar speaker cabinet miking offers many creative recording possibilities.
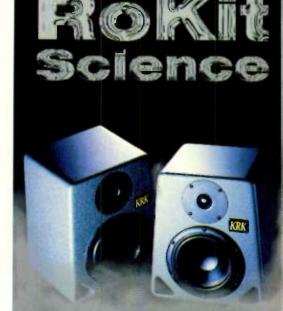
guitar parts. Otherwise the guitar sounds can blend into a confused wall of sound. Bracketing the sounds using tunable high- and low-pass filters (such as those on the Focusrite Platinum Tone Factory or the Drawmer DS201 gate side-chain), can help confine the sounds to narrower parts of the spectrum.

• If you have enough spare tracks, compile a 'best of' solo from multiple whole takes recorded on separate tracks. Hard disk recording is good for this as you simply cut and paste the sections you want to use. However, you can also compile on tape by bouncing the chosen sections to a new track. Compiling before adding delay or extra reverb will help conceal any edits.

• Create a sense of stereo space by processing a mono guitar sound via a gated or ambience reverb program. A close-miked or DI'd guitar part contains no spatial information, but adding reverb to the sound to create the illusion of space may not be artistically appropriate. Ambient reverb settings will simulate the early reflections of a real space without adding reverb decay. A short gated reverb may also be suitable for creating a more live sound.

• In the recording studio, it's common to leave effects processing until the final mix so as to allow sounds to be

changed right up to the last minute. However, guitar players rely on many of their effects to create the right 'feel' at the playing stage — specifically, effects such as overdrive, wah-wah or delay. Reverb can be added at the mixing stage, as stereo digital reverb is sometimes more appropriate than the mono spring reverb built into some guitar amps. If stereo reverb is to be added during recording, a pair of tracks will be needed for recording. Should you want to replace any of a guitarist's 'essential' effects at a later stage, arrange it so the player can monitor via the effect, even if you're recording without it.

• Don't assume the speaker simulator in your preamp is the best one for the job. Better results can often be achieved by taking the unfiltered output from a recording preamp or effects unit, then processing it via a good quality stand-alone speaker simulator. Extra EQ from the mixing desk or an outboard parametric EQ can also help shape the sound.

• Following on from the previous tip, it's often interesting to try splitting the signal into pseudo-stereo via two different speaker simulators — for example, using the simulator in the recording preamp and, at the same time, taking the unfiltered output via an external speaker simulator. Panned left and right in the mix, this can create a larger-than-life sound

RECORDING GUITARS

▶ that is more effective than either of the individual speaker simulators used in isolation. As an alternative, try miking a small amp while also DI'ing it via a speaker simulator. Moving the mic further from the amp will capture more room sound. Combining a close mic or DI feed with a more distant, ambient mic can also yield interesting results. The more distant the ambience mic, and the greater its level relative to the close mic or DI feed, the further back the sound appears in the mix. Compressing the ambient mic output can also help create a bigger sound in a live room.

• You don't necessarily need a big amp to achieve a big sound. A small practice amp can sound great. In addition to the miking arrangements outlined in the previous tip, try putting the mic at head level so it 'hears' what you hear. Also, try miking the side or rear of the speaker cabinet to see what that sounds like. It's easiest to find the best spot if you wear enclosed headphones and move the mic around while the guitarist plays. Lift the guitar amp or speaker cabinet off the ground to reduce bass or stand it right in a corner for more bass. If the sound is too brittle, point the amp into the corner and mic it from behind. It's also worth trying different mics, both dynamic and capacitor, to see which one produces the best tone.

• Compression is a useful tool to even out the tone of the guitar and also to add sustain. By using compression, you may able to get a better lead tone with less overdrive. For clean sounds, introduce EQ after compression: for more mellow results, EQ before you compress. Using compression after gentle overdrive allows more control over the amount of distortion via the guitar volume control without the overall level changing too much.

• Hedge your bets by recording a clean DI feed (via a high-input impedance DI box) on a spare track so you can reprocess it later. This way, if the original sound doesn't work out, you can play the clean track back via a specialised guitar preamp/effects unit or even play it via a small amp and then re-mic it. Alternatively, use both the original and the reprocessed sounds to create an interesting stereo effect.

• When you need a thicker sound, try real double-tracking rather than ADT (Artificial

There are now a number of guitar preamps on the market, such as this Sansamp PSA-1 from Tech 21, which will allow DI recording of authentic 'miked amp' guitar sounds.

"If you play in the control room with your amp in the studio, you can hear what the recorded sound is really like…"
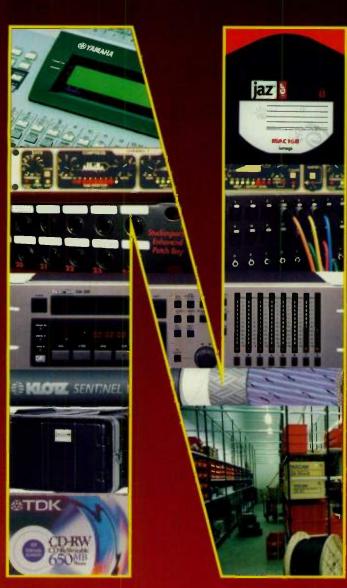
Double Tracking). In other words, play the same part twice over on two different tracks. Depending on the player, you may get better results by muting the original part until the new part has been recorded. If real double-tracking is too difficult, use a pitch-shifter to add a small amount of delay and detuning to fake the effect more convincingly than chorus.

• When DI'ing, you can still use a small guitar amp to monitor what you're playing. This often makes playing seem more natural and the acoustic coupling between the speaker and guitar strings will add life to the sound. Even a small battery-powered practice amp can help you deliver a better performance.

• To get a more lively electric guitar sound when DI'ing or recording with the amp in another room, mic up the strings and add that to the main sound. Use a mic with a good high-end frequency response — a capacitor or back-electret mic is best — and position it around 15 to 20cm from the strings.

• If using a valve amp with speaker simulator, be sure to use a simulator model with dummy load if the amp needs to be silent when you're recording. This is especially important as the output transformer can be damaged by running

with no load. In the case of transistor amplifiers, running without a load shouldn't cause problems, unless the amplifier has a transformer output stage (rare in transistor amps)
If in doubt, check the manual.

• If you play in the control room with your amp in the studio, you can hear what the recorded sound is really like via the control room monitors as you play. However, you lose the acoustic coupling that you get with a loud amp close to your guitar so the sound may be different, especially if it is heavily overdriven.

• When using cabinets with more than one speaker (for example, four by twelves), listen for the best-sounding speaker and mic that one. Miking close to the centre of the cone gives the brightest sound, while moving towards one edge produces a more mellow tone.

• If you decide to use a gate to reduce noise or interference, put the gate after the overdrive stage if possible, but before compression or delay/reverb-based effects. This is so the gate won't cut off your reverb or delay decays. Adjust the decay time so as not to cut off notes prematurely and set the threshold as low as you can without allowing noise to break through. Either an expander or a dynamic noise filter will do the same job, often with less noticeable side effects than a basic gate.

• To get a 'glassy' clean sound, compress the guitar signal and then try adding a little high frequency enhancement from an Aphex Exciter or similar processor. When trying to achieve this kind of sound, DI techniques often work better than miking because more high-frequency harmonics are preserved. If you like a really glassy top, then try switching any speaker simulation out when using a clean sound. SOS

follow that 808!

ROLAND SP808 GROOVESAMPLER SAMPLING WORKSTATION

The new SP808 has little in common with its '80s namesake, the TR808; one was an analogue rhythm unit, the other is a feature-packed sampler, multitrack digital audio recorder, synth and remix tool. Clearly, Roland are hoping that some of the 'hi-tech classic' status of the TR will pass to the new SP. CHRIS CARTER certainly thinks it has what it takes...

When rumours of the SP808 started spreading last year, everyone knew something special was in store. Why? Because there must be some significance in Roland's resurrecting the legendary 808 title after 17 years. Although the accompanying photos of the SP808 go some way towards capturing its 'mean and moody' appearance, they still don't quite do justice to the physical experience of seeing one 'in the flesh', especially for the first time. While it isn't particularly large, when in its 'ready' state it has a certain brooding presence, all black and red, with glowing buttons, just waiting to do its stuff. If its sibling, the Boss SP202 Dr. Sample, looked like a Star Trek Tricorder, as I suggested in my *SOS* January '98 review, then the SP808 is NASA mission control.

Roland call the SP808 a 'Groove Sampler' and are promoting it primarily as a professional DJ/Remix phrase sampler, but it's also an 8-track hard disk audio recorder and a lot more besides. A major feature of this very groovy sampler is its ability to offer fast resampling of phrases and audio tracks, with or without programmable effects. There are 16 assignable sample pads (and 64 pad banks), time-stretch, automatic bpm calculation, an array of audio in/outs, MIDI, an informative backlit LCD, three assignable real-time control knobs and a built-in Zip drive. Also included is a dual D-Beam Controller for live control of effects and sample playback via hand and body movements. Oh yeah, I mustn't forget the onboard programmable 'Virtual Analogue Synth' (VAS) and analogue style 16-step sequencer. Phew! What a line up!

VIRTUALLY RAM

One major difference between this sampler and any other you might have come across is that (apart from a small, unspecified amount of buffer memory) the SP808 includes no RAM. All sampling, recording and song arranging is written directly to Zip disk, using what Roland call Virtual Memory Sampling. This works fast enough and, on the whole, behaves just as if you were using RAM; it also means you effectively have a sampler with 100Mb of memory, without having to fork out extra dosh for RAM upgrades. Nice! Of course, 100Mb doesn't look quite

as impressive for multitrack hard disk recording, but it's still perfectly usable for individual songs.

THE TOUR

The top panel is divided into three sections and each of these is sub-divided again, giving about seven or eight specific areas. Central to the SP808 (physically and operationally) are the Edit, Cursor and Transport sections, which use illuminated buttons throughout. The Edit group allows access to various parameter and editing pages for Song/Track, System/Disk and Sample/Bank. There are Quick Edit buttons for sample Trim, Level, Stretch, Pitch and for Track Play List Region In/Out and Mark On/Off. Also here are the Undo/Redo and the FX Info/Edit button, which takes you to the numerous effects pages and menus (see the box on the available DSP algorithms). Lower down are the Exit/No, Enter/Yes buttons and four cursor buttons for moving around the display and zooming in/out (which double as track editing buttons for the Cut/Paste/Erase/Insert functions). Below these is the large Value/Time rotary dial, Shift and Vari-Pitch/bpm buttons. The Vari-Pitch function affects the overall playback pitch of everything — samples and audio tracks — although at the standard 44.1kHz sampling rate it will only lower the pitch, whereas if the SP808 is set to 32kHz sampling it will also raise the pitch. The song bpm can be adjusted 'on the fly' from 50% to 200% of the original, but this only affects the playback speed, not the pitch.

The left side of the control area handles audio — feeding it in, manipulating it and and feeding it out again. Top left are input controls for Mic, Line and Aux and headphone level. Below this is the small, but informative, backlit LCD. The top half of the display always shows, in this order: the current song position (measure/beat/tick, switchable to hours/mins/secs/frames); the current bpm; the sample rate of the Zip disk; and the current song number and sample bank. The lower half of the LCD changes, depending on the current mode and edit status, and can also show the extended (and aptly named) Big Time display. Many of the edit screens can be scrolled downwards or sideways and some can be zoomed in or out. The default display on power-up is Meter, which shows six stereo bargraph meters; one for each track, one for the external inputs and one for the master output (See Figure 1). Below the display are four illuminated buttons for the song locator, and using the shift button, eight locator points can be set. Once these are placed, you can instantly move to any set point in a song by pressing the relevant button.

These are also used for setting automatic punch-in/out points when recording. Next to these is the display Play button, which allows you to call up the four basic display screens (Main Meter, FX-Aux Meter, Play List, Big Time) from any other screen or sub-display — useful in panic situations when you might be feeling a bit lost! There are also two audio preview buttons, To/From and Scrub. Used in conjunction with the Value/Time data wheel, these allow you to repeatedly play short segments of a selected track to help when editing or adjusting sections of audio, and are similar to using a jog/shuttle dial on a video machine. The lower left-hand side is where the business of recording and mixing goes on. Each of the four stereo tracks has a tricoloured status button: Mute (off), Play (green), Record (red), Bounce (orange); an illuminated Effects Send button (pressing this brings up the display for channel EQ, FX send etc.); and finally a stereo channel fader. There is also a fifth, stereo channel, switchable to Mic/Line or sample trigger pad input, and with the same type of effects/EQ button as the others. This channel is used for adjusting the overall level of the sample pads or external inputs into the audio recorder. Finally, there is the single fader and illuminated mute button for the master stereo output.

The right-hand side of the SP808 is where you find the D-Beam real-time effects controller (see box), the Effects On/Off button and the Step Modulator On/Off button (see box). But pride of place on this side must go to the quaintly named Sample Palette. As on the small but perfectly formed Boss SP202, Roland have gone for the strangely eerie, large translucent-glowing sample pad look. In this case though, there are 16 of the babies, and a nice touch is that they automatically load up any samples (and glow) when a Zip disk is inserted. The pads are also used as character keys for entering names for songs, banks, and effects patches, but not sample names. This was one of the few disappointments in the SP808 — sample numbering rather than naming. There are four other buttons: Sample, Sample Bank, Hold/Tap (also used for bpm calculation) and Clipboard (more on this one later).

FAST AND FURIOUS

Roland have made sampling on the SP808 about as fast and as easy as it gets. You can stick with the default settings or set your own sampling parameters quite easily, as the display jumps to the sample options screen whenever you press the Sample button (see Figure 2). Sampling options here include: ▶

SAMPLE/RECORDING TIMES

Virtual Memory Sampling, using R-DAC compression.
Internal processing: 20 bit,
S/N ratio: 92dB.
Freq response: 10Hz to 21kHz
(@ 44.1kHz).
Maximum time per Zip disk: 23 minutes stereo @ 44.1kHz; 32 minutes stereo @ 32kHz; 46 minutes mono @ 44.1kHz; 64 minutes mono @ 32kHz.
Samples: 1,024 (16 pads x 64 banks).
Polyphony: Stereo x 4 (samples, audio tracks or any combination not exceeding 4).
Songs: 64 (with 2,000 'phrase events' per song).

Roland SP808

STEP MODULATOR

This is a software approximation of a pre-MIDI, analogue-style step sequencer (specifically the System 100M 182 module). It can be used in conjunction with the VAS, or to control designated parameters on any of the DSP effects. Pressing the Step Mod/Set-up button brings up a display showing two rows, each with eight vertical bars (see the LCD screenshot in this box). Each of these bars represents a single note, and the higher the bar, the greater the parameter value. Using the cursor buttons, you can navigate around the display adjusting the notes with the Value dial (even while the sequencer is running). A small box on the left shows the value of the selected step, which changes from numbers to notes if editing patterns for the VAS. Pressing the Step Mod button starts and stops the sequencer, and a small arrow steps along the display at either the bpm set by the sequencer or current song. Other editable parameters are: Step Time, Gate Time, Rest, Trigger (Single, Repeat etc.), End Step, Series (16) or Parallel (2x8) and Copy (for copying patterns between patches). This brings me to the only real problem with the Step Modulator which is that each pattern is tied to an effect patch. There is no separate memory for keeping just sequencer patterns in. If, for example, you want to use a number of different sequences with the VAS, you need to make a new VAS patch for each pattern and step through them in the effects bank. This methodology shows us how Roland see many features of the SP808 being used to make separate sample phrases for dance music. Using their technique, you sample each sequencer-controlled VAS patch into a pad, and play them back as sample loops, which actually works surprisingly well. It just takes a little longer to get there than it normally would by selecting various sequencer patterns to control a single synth sound.

MAXI MIDI

The MIDI specification for the SP808 covers seven pages; it can sync with other MIDI machines as master or slave, using MTC, MMC and MIDI clock, sample banks and effects can be switched with ease, and the D-Beam can act as a useful MIDI controller. Editing or changing effects and sample settings via MIDI SysEx is not supported, but you can use MIDI controller messages for most of the digital mixer functions.

▶ Stereo/Mono, Input Source, Sample/Effects Options, Trigger Type (manual, by pad, by level, play button), Auto Divide, Auto Trim, Loop Type, Pre-Trigger (0-320 milliseconds), Available Sample Time and an Input Level Meter. If you are recording from a commercial sample CD, you can just grab sections of audio including any pauses or spaces in between samples, and Auto Divide will trim out the empty sections and place the new individualised sample loops into consecutive empty sample pads — sometimes the loops may need tweaking, but on the whole this is a useful time saver.

As if to emphasise that SP808 is a sampler, the Sample button glows bright red and has a thick red

outline around it, just so you don't miss it! This really is easy-peasy sampling — assuming you are happy with the sampling parameters, you just tap the Sample button and any pads containing samples are dimmed and the the first empty sample pad available begins to glow, so you can see exactly where your sample will be placed. When you actually want to begin sampling, you tap the Sample button again (the display then shows how much disk space you are using). To stop sampling, just tap it once more. The Enter button then flashes green (at this point you can tap any sample pad to hear your new sample). If all is OK, press Enter and the new sample is placed into the relevant pad position. If not, press the Exit/No button, and the display asks if you wish to try again. If your sampling was successful, the Sample button begins flashing again, a new sample pad is automatically chosen ready for the next sound, and off you go again. This all happens in a matter of seconds and is actually a *lot* faster and easier than it sounds on paper! If you try sampling into a fully occupied bank, all the sample pads and the Enter button begin flashing and the display asks you to select a pad. You can choose to overwrite an existing pad or choose another bank containing empty pads, and given that there are 64 banks each containing 16 pads, not being able to find an empty slot should be a rare occurrence.

A LITTLE SNIP...

Roland have tried to make editing samples as straightforward as possible too, with special 'Quick Edit' shortcut buttons to place you directly into the ▶

RETRO DSP

The DSP Effects bank is quite a powerhouse and Roland have taken the retro/analogue theme further by including some amazingly faithful algorithms of classic Roland rack effects and Boss effects pedals of the last 20 years. Whenever the FX button is illuminated, the three real-time control knobs become active, and if the FX INFO button is clicked, the display shows a graphical representation which updates in real time as you twist and turn the knobs. A Select Row button decides which parameters the knobs will control and the DSP effects can function in a number of different ways, with the 'Isolator' and 'Filter' modes being the most accessible and called up by simultaneously pressing Select

Row and Shift. You can instantly jump back to these modes from any other filter effect, but although the two effects blocks are connected in series they can't be used at the same time. The 'Filter' is a basic 24dB/oct low-pass type with a 'Low Booster' and real-time controls for Frequency, Resonance and Boost Level, while the three-band 'Isolator' can reduce high, mid or bass audio bands by up to 60dB — useful for stripping out elements of a sample or track. Both these modes are also available (simultaneously) in the effects bank as a combined algorithm (01) with many more editable parameters. The effects list (see box) gives some idea of the basic processes available, although what isn't so obvious is that many of the algorithms contain more than one effect

block. For instance, the Stereo Dynamics Processor contains a comp/limiter, an enhancer, a three-band EQ, a noise suppressor and more than 20 programmable parameters. The Reverb/Gate also includes a three-band EQ and more than 30 editable parameters. Generally, the effects are top-rate; the only negative points I can find are that effects are applied globally to the sample pads (no individual sends) and you can only use one effects algorithm at a time. There are ways around these restrictions, as you can resample a pad (or pads) through the effects bank, and then resample again, adding another effect if need be. Using this method allows you to have a different effect on each and every pad while using the effects bank for the mixer and audio tracks.

Roland SP808

▶ editing pages with just one or two key clicks. These shortcuts are incredibly useful — click the Trim button and, for any currently active/playing pad, the display shows a list of edit parameters: Sample bpm, Start Point, Loop Point And Length. Scroll up or down the page and you can view Play Mode, Loop Mode, bpm Base Note, Mute Group and Level (you can also view the sample level by clicking the Level shortcut). Press another sample pad and the display updates to show the parameters for that sample instead. Working this way, you can jump from sample to sample, trimming and looping without leaving the basic edit page. Fast? You bet. The only let down here is that all sample editing is carried out numerically and by ear — no waveform editing. This is annoying because the SP808 seems to be capable of displaying sample waveforms, as it does so when you use the preview Scrub function, so maybe it will be introduced with a later system update. As with most bpm/groove-based phrase samplers, editing loops is a lot easier if you have a good ear for rhythm and get the loop as close to perfect as possible when you sample it (practice make perfect). The SP808 is fast and responsive in this department, and I must admit I found it quite easy to get almost perfect loops most of the time in the sampling stage. It also helps a lot if you know the bpm of the sound you are sampling, because, once in the edit page, you can adjust the start and end loop points while keeping an eye on the sample bpm figure (which updates as you edit) until you get the perfect loop. Alternatively, you can set a loop by playing the sample and tapping the Enter button at the loop point. Not the best solution, but useful in certain situations.

When used with the shift button, the Trim and Level shortcuts also take you directly to the Stretch and Pitch editing pages. On the Time Stretch page are parameters for: Source Pad, Target Pad, Ratio (50 percent to 150 percent, where 100 percent is the original length), New bpm, Match With and Match Type. Basically, you select a sample pad to stretch, select a pad to place the new stretched sample into (or overwrite the original), dial in a new bpm or ratio (it doesn't matter which, as the figures are relative to each other), click the Enter button and... voila! You have a surprisingly good-quality stretched sample, in mono or stereo. If you're not sure of the bpm the SP808 will match the new bpm/Ratio to an existing loop by tempo or by sample length. When matching to length you can specify that the new sample will be proportionally longer or shorter by a fixed ratio (plus or minus x2, x4, x8, x16). The Pitch function works in a similar manner to Stretch, but instead has parameters for: Grade (5 types); Pitch (+/-12 semitones); and Fine (+/- hundredths of a semitone). It's useful for changing the pitch of loops while keeping the bpm untouched, but the quality suffers more than with the Time Stretch feature, and samples can end up sounding very 'flammy' (and at extreme values, like they are running through a delay effect). Unfortunately, there seems to be no other way of adjusting the sample playback pitch, so you can neither play samples chromatically across a MIDI keyboard (although you can trigger them) nor apply MIDI pitch effects such as modulation or pitch-bend. Shame. Other sample editing functions, such as Delete, Copy, Reverse, Renumber, Divide, Name Bank, Copy Bank, Delete Bank are accessed via the Edit Sample/Bank menu.

The Clipboard is brilliantly simple but, at the same time, incredibly versatile. Simultaneously press a sample pad and the Clipboard button and the sample is transferred to the clipboard. At this point, the Clipboard button acts just as if it were another sample pad and you can play samples and phrases with it. Pressing Clipboard with another sample pad transfers the sample and holding down the Shift button at the same time sends a copy to the new pad, leaving the original sample in the clipboard. If you've recorded any audio tracks, you can highlight a section of audio using the Mark button and by simultaneously pressing Mark and Clipboard you can copy audio tracks to the clipboard. These can then be transferred or copied to a sample pad for editing and manipulating as if they were samples.

TRACK EVENT RECORDING

Songs can be constructed in two different ways. The first method, called Track Event Recording, functions by recording a playlist of trigger information from the way the pads are played, in a similar manner to using a MIDI sequencer. Effects, EQ, multiple sounds and velocity are not recorded (unless the pad is triggered from an external MIDI source), just the pad timing/trigger information. This isn't actually as much of a restriction as it sounds, because you can layer samples across all four tracks and quite complex arrangements can be constructed. The main benefit

of using Event Recording is that it uses very little memory and makes it easier to precisely edit and rearrange individual samples/phrases after they have been recorded. Event Recording also allows you to lay down samples and loops in Step Time by tapping pads and entering parameters for Measure, Step Time, Duration, Volume and so on. This can be handy for recording a lot of repetitive loops or arranging particularly precise or fast rhythm tracks. If you leave one track free while Event Recording, you can digitally bounce down three Event tracks onto a stereo audio track, adding effects and EQ as you go, and possibly mixing a live performance through the external audio inputs as well. Using this method, you can bounce down mixes without any loss of quality. Event Recording isn't quite as versatile as using a MIDI sequencer, but it works just fine for basic song construction (and deconstruction).

The other method, Track Audio Recording, allows you to record sound as you would on a multitrack tape machine or hard disk recorder. The one restriction is that although this is an 8-track recorder, you can only record tracks in pairs, and only one pair at a time. You are allowed to specify if a track is mono, but this only increases the available recording time, not 'polyphony' — you can only ever play back four stereo/mono audio tracks or samples. All the expected transport controls are present: Stop, Play, Record, Rewind, Fast Forward, Return To Top, and Measure Step Forward and Backward. Most of these buttons are dual purpose (using the Shift button) and some perform different functions depending on whether you are working in Event or Audio mode. All audio recording is accomplished using the five-channel mixer, and there are numerous recording configurations available: external source (mic or line); the sample pads; digital bounce down; bounce down and pads; bounce down and pads and external input; all with or without effects. Once the tracks contain audio, a wealth of editing options are available. You can cut, copy, paste, erase, insert, move across tracks, shift the timing, and send audio to the pads to be edited as samples. Considering the size of the LCD, a very useful moving 'piano-roll' display shows you the tracks and audio segments of the current song, and you can zoom out for an overview or zoom right in for greater detail (see Figure 3). Each of the five channels has its own three-band parametric EQ, Merge L&R, Effects Send (pre or post), Aux Send (pre or post) and Pan control. The stereo Aux In/Out can also be configured as a master effect insert for the main stereo output (pre or post the main fader), for further EQ or compression from an external unit. If the optional OP1 Multi I/O board is fitted, you have the ability to send mixes to DAT, MiniDisc or CDR via co-ax and optical digital outputs, or to another mixer, using the multiple stereo track outputs. All in all, a very comprehensive and professional setup.

Some of you may be thinking that four stereo tracks and four-note polyphony doesn't sound that fantastic, but these restrictions aren't too much of a worry because one of the most useful features of the SP808 is fast and easy resampling of recorded audio to the sample pads, and vice versa. Add to

SP808 OP1 MULTI I/O

Having used the optional OP1 Multi I/O interface, I would say it's a pretty essential purchase. What you get, in addition to the already well equipped rear panel, is an external SCSI socket, six track outputs (on three pairs of phonos), two sets of digital I/O (co-ax and optical) and the ability to send and receive digital audio at 44.1kHz. Also, if you use a CD player with a digital output, you can sample entirely in the digital domain and not have to worry about setting sampling input levels. The extra track outputs come in handy for feeding tracks to an external effects unit or mixer. The SCSI connector is a bit of a mixed bag though. Seeing the standard type 25-pin socket I was quite looking forward to trying out different hard drives with the SP808 but alas this connector is *only* compatible with Iomega Zip drives, not even Iomega Jaz drives. I suppose there must be a good reason why Roland have restricted the usefulness of this connector, but I can't for the life of me think what it might be. Jaz drives, in particular, are falling in price still further, even as I write, and would make a perfectly good partner for the SP808. Maybe this shortcoming can be addressed in a future system update. In the meantime (particularly if you are going to use the SP808 as a multitracker and need to back up your work), I would definitely recommend including the OP1 Multi I/O and an external Zip drive in your budget.
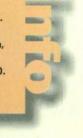
this resampling equation the real-time effects manipulation, EQ and the Virtual Analogue Synth, and you begin to realise what power you have in your control. You can quite literally keep resampling endlessly, back and forth, through effects and EQ, changing them each time, if you wish, adding more samples or loops, and recording your results on to an audio track without leaving the digital domain at any time.

AND FINALLY...

Listening to A/B comparisons with digitally sampled material (at 44.1kHz) I found the sound to be almost faultless, and this is in spite of the fact that the SP808 uses compression (R-DAC). I could have used the SP808 for a number of remix projects recently, I just wish it had turned up sooner because I would be quite happy producing finished remix masters with it. But the SP808 can be a lot more than just a professional remixing workstation, and because it's relatively affordable, many more potential users will be attracted to it — home and project studio recording, jingle production, spot effects, schools and theatres. There's no doubt that the SP808 is operationally complex, and even though it is awash with controls and visual aids, it's not as immediately accessible as some sampler/audio workstations. Still, most purchasers will be able to get usable results within a couple of hours and I am sure that anyone investing a grand or more on such a 'pro' tool as this will be willing to spend quite a few intensive sessions with it to get to know its true capabilities. This is one hell of a machine. It's going to be massive; I certainly want one. Top marks to Roland for coming up with what is sure to be another 'classic 808'. **SOS**

DIMENSION D

As already seen on the Roland MC505 (SOS April 1998) the D-Beam controller (licensed from Interactive Light) is ideally suited to the SP808. This is a hands-off controller, used in a similar way to a theremin, but utilising entirely different technology. Two invisible beams of light are projected upwards, and by waving your hands over the top of the D-Beam you get some form of control (occasionally unpredictable) over various sampler functions. There are three preset buttons under the D-Beam: Pad Trigger, Pitch and Effects. The Pad Trigger lets you select two pads for beam control, Pitch allows extreme pitch changes (but only in a downward direction) and Effects gives you control over specified effect parameters (the default settings are for 'Isolator' and 'Filter'). A great D-Beam set-up is Effect Preset 99 'AnlgSyn'. This allows you to use the D-Beam and real-time controllers to 'play' the Virtual Analogue Synth. A scream. Literally!

World Radio History

World Radio History

ebb&flow

BON HARRIS • PROGRAMMING FOR SMASHING PUMPKINS

SAM MOLINEAUX discovers an early techno pioneer thriving in the heart of Hollywood; living proof of the maxim 'what goes around comes around'.

ack in 1983, five years before the official birth of techno in the clubs of Detroit, a couple of Essex teenagers started fooling around with modular synthesizers to create their own brand of minimalist, beat-driven electro music. They called themselves Nitzer Ebb (an entirely made-up name) and, eschewing the disco-friendly synth pop that was in vogue in Britain at the time, turned for inspiration to bands such as Bauhaus, The Birthday Party, Killing Joke, DAF and Die Krupps. With their futuristic outlook and combination of post-punk rock and forthright electronic rhythms, the early Nitzer Ebb became somewhat unwittingly associated with the emerging European industrial dance scene. However, thanks to a string of consistently precocious albums coupled with relentless touring (including worldwide stadium tours with Mute labelmates Depeche Mode), they eventually moved into a class

of their own, deliberately at odds with most of the popular music of the day, electronic or otherwise.

The UK music press, increasingly lost for words to describe them, rewarded Nitzer Ebb with little more than indifference. But outside their country of origin it was a different story. Throughout the late '80s and early '90s, Nitzer Ebb enjoyed a huge underground following, both in Europe and in the prospering US club scene which welcomed their synth and percussion-led techno-rock.

Their first two albums, *That Total Age* and *Belief*, are still hailed as seminal dancefloor classics. The subsequent *Showtime* and *Ebbhead* saw them embrace a less rigid, funkier synthetic approach. By the time of their fifth and final album — 1995's sadly inaccurately titled *Big Hit* — they'd begun to experiment with a combination of live instruments and electronics, both to broaden their sound and to add interest to their live performance. At this point,

the wheel. He's working on various film and video game soundtracks (see the 'Game On' box), and has a new recording project, Maven, which in many ways picks up where Nitzer Ebb left off. It's a more organic hybrid of synths and guitars (provided by John Napier of LA industrial/dance band Ethyl Meatplow), but this time with Harris on vocals.

"There's still a very strong electronic edge to it, but I'm also trying to interweave live played elements with the electronics. I'm not so much into just having a studio band these days, so this is definitely going to have a strong live element, with a full band and a full performance. I really believe that the more you can have that's live, the more true the experience is. There are a few obvious similarities to Nitzer Ebb, simply because I was a central part of that band and I gave it everything I had, but the Maven material is a different side of my vocal style than anyone's ever heard."

With nearly an album's worth of songs in demo form and currently two very interested labels, Harris is hoping to start recording in Los Angeles before the summer's out. And it's likely that Flood will be at the helm — the highly unconventional, not to say enigmatic producer, who was responsible for all but the first of the Nitzer Ebb albums.

"We're both really busy so there's always a chance that that might not happen, but he digs what I'm doing and I dig what he does," says Harris of their potential collaboration.

PUMPKIN EATER

In the meantime, both Harris and Flood have both been working with Smashing Pumpkins' Billy Corgan — himself a Nitzer Ebb fan and a fellow believer in breaking down the barriers between musical genres. When Harris and Corgan bumped into each other at a party just before Christmas, they got talking about a possible collaboration.

"Billy was trying to do something different, to take the Pumpkins in a different direction," recalls Harris. "Towards the end of the last record (the Grammy award-winning *Mellon Collie And The Infinite Sadness*) he was delving into technology and getting into the synth world and he was saying he wanted someone with a little more experience to come in and just see what their take would be on what he was doing. So he offered me a couple of tracks which I went away and reinterpreted for him from a very synthetic standpoint. I took them back to him and he really liked what I done."

Consequently, Harris was drafted in towards the end of the sessions for the Smashing Pumpkins' new album *Adore*, was provided with his own studio space at Hollywood's Sunset Sound ▶

"So many artists have suddenly decided they're going to 'go electronic'. But it's not really an integration, it's them singing over the top of an electronic track."

the band had finally given up on the UK and relocated to the States. But tensions during the recording of *Big Hit* and the follow-up tour eventually took their toll and the band split up.

These days, vocalist/guitarist Douglas McCarthy is back in England lending his voice to Recoil, the pet project of Depeche Mode's Alan Wilder. Meanwhile programmer/percussionist Bon Harris is living in Los Angeles and experiencing something of a career renaissance thanks to the popularity of what has become known there as electronica (see 'La Musica Electronica' box).

"People see us as one of the originators of that scene, and we're getting a lot of props from being around way before" says Harris, who admits that, although he is delighted with the current US interest in electronic music, he has a bit of trouble understanding what all the fuss is about.

"It's certainly nothing new, but I suppose like life, music tends to go in circles and things come back into fashion."

Harris' musical career is also undergoing a turn of

LA MUSICA ELECTRONICA

For anyone unfamiliar with the term, 'electronica' is the all-encompassing US term for the latest wave of synth and dance music that exploded on to the American music scene last year, effectively stamping out the last dying embers of guitar-based grunge rock.

Although America has had its own flourishing electronic dance music and hip-hop scene for years (mostly centred around cities such as Chicago, Detroit and New York), it has really only been in the last 12 months or so that it's moved up a gear and entered the mainstream — and then largely due to the influence of British groups like Prodigy, The Orb and the Chemical Brothers. Now even the formerly rock and pop-dominated West Coast has opened its arms to techno, rave, trip hop, jungle, drum & bass, big beat, and all their various offshoots.

"It's been a while coming, but it has been percolating underground for some time," notes Harris. "The first time we came over to tour, which was in 1989, there was a very strong scene, but it wasn't massive. So it's been going on a while, but now it's seeping into the mainstream where you're getting commercials and movies with very heavily electronic soundtracks. It's definitely rising to the surface."

▶ and given the task of adding a new electronic edge to the Pumpkins' sound.

"Everything was backed up in *Pro Tools* so they'd burn me a CD of *Sound Designer* files of whichever tracks I wanted from the multitrack — normally I'd just have vocals, bass and drums — and Billy would just say 'Go for it, whatever springs into your mind, just do'. So using *Logic Audio* to sequence, I was basically bedding my stuff underneath and around what was already there. As I finished each song, I'd record the analogue sequences as one long track, burn those down on a CD and give them back to Billy."

For his sound arsenal, Harris relied almost entirely on modular synthesizers: his trusty old Roland System 100M and an Oberheim Xpander, along with a modern Doepfer A100 and the recently-launched Clavia Nord Modular.

"Much as I like working with old modular synths, you can pretty much only do one sound at a time. You have to record that sound immediately and do a patch sheet. It's a very old-fashioned and time-consuming process. Things like the Nord Modular make it a little more feasible to do it with a deadline".

VINTAGE + DIGITAL

The only purely digital instrument Harris has in his setup is a Kurzweil K2000, which he uses primarily for its sampling and DSP capabilities.

"With the Kurzweil, you can take virtually any sound, bend and distort it and do whatever you

The Smashing Pumpkins (with Billy Corgan, left) and their new album, Adore.

Something Old, Something New (Part I) — Bon's giant Roland 100M modular system shares studio space with the all-new Clavia Nord Modular, a laptop PC, and a top-flight Power Mac running Digidesign Pro Tools.

want, until it's no longer recognisable. It was a good system — vintage analogue synths on the front end and then quite sophisticated digital editing and digital processing on the Kurzweil on the other. It provided a lot of possibilities for manipulation."

Harris worked on just over half the songs on *Adore*, subtly reinterpreting certain areas and adding some drum programming in places, but never to the extent of detracting from the Pumpkins' alternative rock style. Nevertheless, anyone familiar with Nitzer Ebb will certainly notice Harris' influence, particularly on the album's first single 'Ava Adore' with its solid synthetic pulse. Presumably Billy Corgan was happy with the results?

"He pretty much liked everything I did and he seemed very happy most of the time," says Harris. "But there were so many slave reels for each of these songs, so many guitar and drum parts, it was inevitable that the mix process was going to involve a lot of weeding through the things. We both knew that, great as some of the ideas were, they might not make it into the final mix just because there were so many other good parts vying for position. Some of the songs are very acoustic and stripped back, and didn't really warrant my input at all."

During the recording sessions, this combination of quite heavily electronic and almost entirely acoustic songs seemed a little bit at odds for one ▶

GAME ON!

Recently, with video games achieving acceptance with a more mainstream and less nerdy audience, manufacturers such as Sony, BMG Interactive and Activision have started looking to more established composer/producers and current Top 40 acts for their games soundtracks. Sony's new *Gran Turismo*, for example, features cuts from Garbage, Manic Street Preachers (remixed by the Chemical Brothers) and Ash; their new 'psychedelic tunnel game' *N20* is set to a soundtrack of remixes by Las Vegas big-beat protagonists Crystal Method; and BMG have just announced their World Cup Team video game *Three Lions* with music from Ocean Colour Scene.

The potential for licensing existing

music is great and offers lucrative promotional opportunities for the groups involved, but it's also an exciting area to be working in at ground level, particularly for a musician like Bon Harris, who can fill in time between larger projects writing video game soundtracks from his own desktop studio.

Harris has recently completed the music to Activision's new road-rage action game, *Vigilante8* (a sequel to their popular *Interstate '76*).

"I was lucky on the Activision projects in that they can use CD-quality audio which means you don't have to mess around reducing the sample rate or bit resolution of the music you give them; you pretty much just write a song, record it to DAT and they burn it onto the CD from there. It's a condensed

version of working on one of your own album songs, except you're working to a brief from somebody else," he explains.
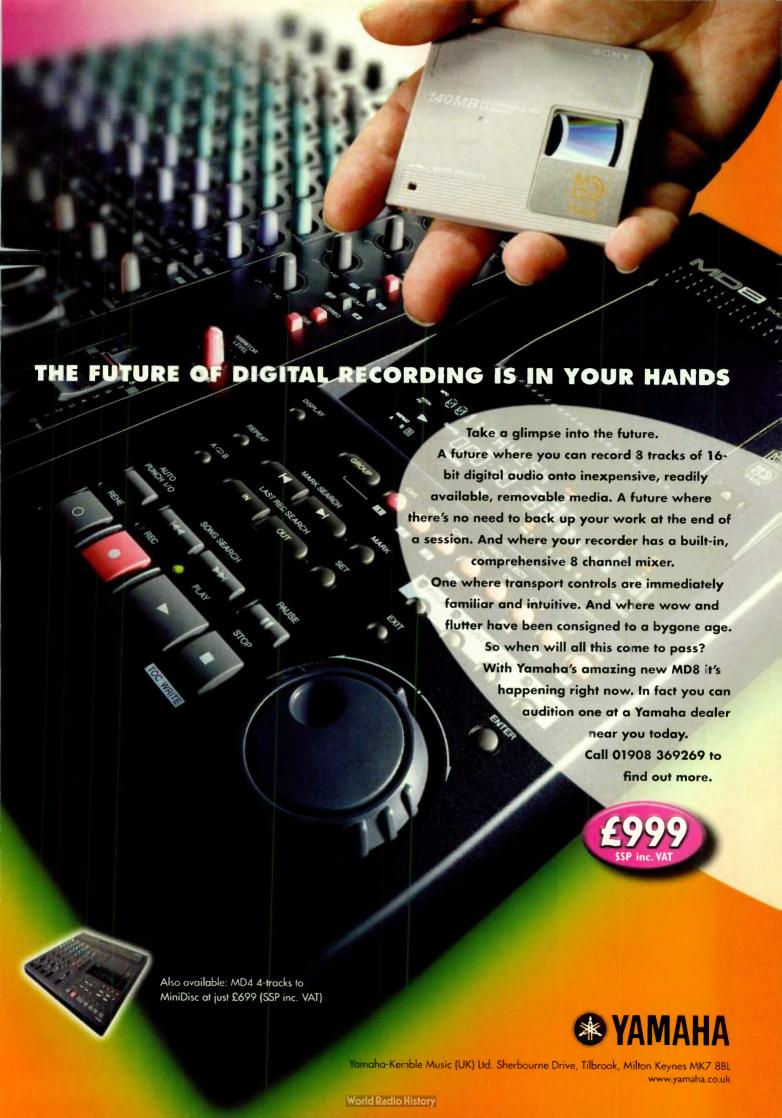
"You get a character description and sometimes a description of the scene: make it fast, make it aggressive, this is the character, and so on. For one of the projects I worked on, they wanted '70s funk music but futuristic, so you just have to interpret whatever the hell futuristic funk music might sound like! I looked around at some of the beats that are in vogue nowadays, and found that there's actually lots of similarities to the older stuff — they're just faster, more processed and more compressed.

"It was actually quite fun and it's interesting when you pull it off. I put a cut-up beats type of framework on the whole thing to give it the futuristic feel, but then added more conventional '70s

wah-wah guitars, brass sections, Hammond organs and the like, which I used a lot of samples for."

Since recently moving into a new production space on Hollywood's Sunset Boulevard, Harris has also begun delving into film music; at the time of this interview he'd just completed a submission for a new Kenneth Brannagh directed movie currently in production.

"Really, whatever comes around, if it's an interesting project that I want to get involved in, I'll work on it. I don't necessarily take a film project 10 times more seriously than a video game. You usually have less time to invest if you're doing a video game because, on stuff like that, the budgets are smaller, but if I'm into it and I think it's a good thing to be associated with, then I'll do it."

THE FUTURE OF DIGITAL RECORDING IS IN YOUR HANDS

Take a glimpse into the future. A future where you can record 8 tracks of 16-bit digital audio onto inexpensive, readily available, removable media. A future where there's no need to back up your work at the end of a session. And where your recorder has a built-in, comprehensive 8 channel mixer.

One where transport controls are immediately familiar and intuitive. And where wow and flutter have been consigned to a bygone age. So when will all this come to pass? With Yamaha's amazing new MD8 it's happening right now. In fact you can audition one at a Yamaha dealer near you today. Call 01908 369269 to find out more.

£999
SSP inc. VAT

Also available: MD4 4-tracks to MiniDisc at just £699 (SSP inc. VAT)

❋ YAMAHA

Yamaha-Kemble Music (UK) Ltd. Sherbourne Drive, Tilbrook, Milton Keynes MK7 8BL
www.yamaha.co.uk

BON HARRIS • PROGRAMMING
FOR SMASHING PUMPKINS

Below: Something Old, Something New (Part II) — on top, the Doepfer A100 modular system, fully analogue throughout, while underneath lurks the entirely digital Pro Tools I/O hardware and Alesis ADAT.

Above: Something Old, Something New (Part II) — A well-worn Korg MS10 sits atop a Sony DAT and Bon's highly prized Kurzweil K2000R, as well as an Oberheim Expander and Roland S50 dating from Bon's days with Nitzer Ebb.

▶ album's worth of material, but, as Harris explains, with Corgan at the production helm and Harris' old associate Flood coming in at the mixing stage it worked out very smoothly.

"It's quite a testament to the mix job that nothing really jumps out at you as being super-different, which was some of the craft of Billy and Flood getting together for the mixes. You're aware of there being a different approach, but no one thing or the other is highlighted. It's very much a subtle integration all round, and it's really forged its own sound, it's not like 'Here we go, this is electronica!'"

SMASHING BOUNDARIES

But as a genre, electronica is already beginning to spill out of any strict definitions. While alternative rock bands like the Smashing Pumpkins welcome synths back into their fold, former techno acts are bringing in real instruments to enhance their live performance. And at the same time solo artist/producers are realising their own rock-pop-electronica fusions. This middle ground is essentially what Harris and McCarthy were investigating back in the last few years of Nitzer Ebb, and it's an area Harris still sees as having great potential.

"So many artists have suddenly decided they're going to 'go electronic'. But it's not really an integration, it's them singing over the top of an electronic track. That's one of the reasons why the new Pumpkins thing is so good, because it's not like that. Similarly, it's what I'm working on with my own project, although I'm coming at it from the electronic angle, and trying to build upon those foundations."

ON THE EBB?

So does all this mean that Nitzer Ebb is no more? Harris: "We haven't disbanded as such, but it's highly inconceivable that anything will happen for a very long time."

Nevertheless, the recent reawakening of interest in '80s synth pop has led to a number of newer electronic acts expressing interest in reinterpreting some of the early Nitzer Ebb material. There's apparently a 'best of' remix compilation in the pipeline (once a legal dispute involving their first two albums and their US and UK record labels is settled). Meanwhile, acts such as Kraftwerk and Gary Numan, both currently on major venue tours of the US, are reaping the rewards of having been there at the outset.

"I think it's great that someone like Gary Numan can still be around and doing his thing. He does have certain enduring qualities, and he's still good at what he does," says Harris. "There are lots of great bands and lots of great sounds around at the moment, especially in the current electronic scene, but I don't see many of them really pushing the envelope of the genre anywhere. There are still not really any songs and relatively few personalities involved. I think Portishead are one of the few examples of a band with personality that uses electronics in an interesting way, and there's Tricky, who's awesome — but not much of the more straight up electronic stuff is actually forging its own sound."

While working at the forefront of a newly flourishing electronic scene has brought its share of new challenges and increased diversity, Bon Harris' musical philosophy remains unchanged.

"Even working in the commercial realm, as I have been lately, I still find myself incapable of doing something totally throwaway. It's got to be real, and there has to be an undeniable emotional presence in what's happening, otherwise it isn't music. All the music I listen to has that streak of sincerity and honesty in it. The moment that you don't feel like you mean it, stop!"

BON HARRIS' HIT KIT

• **OBERHEIM XPANDER**
"This was one of the first professional synths that I bought. Pretty much whatever budget we had for equipment at the start, a lot of that stuff has made it through 15 years of use."

• **DOEPFER A100**

• **ROLAND SYSTEM 100M**
"You patch it together with patch cords, it's very large, not very portable, and it looks like a telephone exchange when you've finished doing it; it only does one sound at a time and is terribly arcane and awkward, but I love it!

"If there's a particular sound I'm after, I can pretty much get it on the System 100M. I've often thought if I was banished to a desert island and told I could only take one piece, this would be it."

• **CLAVIA NORD MODULAR**
"This is a synth I'm using a lot at the moment, mainly because I'm working much more in commercial areas. The thing I like is that when you've finished wiring up a sound with the virtual patch chords on your PC, you send it to the tiny keyboard and then you can just take that under your arm to the studio."

• **APPLE MACINTOSH 9500**

• **EMAGIC** *LOGIC AUDIO*

• **DIGIDESIGN PRO TOOLS**
"When you're using analogue synths, Pro Tools is a good way to go because it speeds up the process somewhat. You can take a minimal amount of input, record that and then start chopping it and further processing it. I've got it hooked up so my K2000 is on the network with the Mac, so anything that I cut from a Pro Tools track can be sent over SCSI to the Kurzweil and processed that way."

• **KURZWEIL K2000 (WITH SAMPLING OPTION)**
"I remember before the K2000 came out, Flood and I would have these 'wouldn't it be great if' conversations, and we were saying wouldn't it be great if you could basically have a modular synth but for samplers, like the early editions of [Digidesign's] *TurboSynth*. So pretty much as soon as the Kurzweil came out Flood and I looked at each other and went 'There's your real-time *TurboSynth*!'

"I also use it as my main sampler — it's a little more complicated to use than an S1000, but I think generally the sound quality is better and there's so much you can do in post-production with it. It reads Akai files, Roland files, Ensoniq files — it's a real workhorse. Most samplers I've had before, until you put a disk in, they don't have any sounds, whereas the Kurzweil's got some really great sounds straight off. That's a big time-saver when you're working on commercials and soundtracks."

Return of the Native

TC WORKS NATIVE EQ & NATIVE ESSENTIALS PC PLUG-IN SOFTWARE

TC Works transfer more of their expertise to DirectX PC software with a set of interfaces that let you squash, tone up, and space out your audio with a mouse. MARTIN WALKER gets clicking.

TC Works certainly mean business. Only a few months have passed since the launch of their *Native Reverb* (mentioned in the PC plug-in roundup in *SOS* March '98), and now they are back with a pair of all-singing, all-dancing EQ plug-ins and a budget bundle of effects, all of which will run inside any DirectX-compatible PC application. The *Native EQ* has two completely separate EQ modules. The first is parametric, providing seven fully configurable bands, and the second is a 28-band graphic (which can also be switched to either 14 or seven bands if required). The *Native Essentials* bundle contains a trio of useful tools: *Q* (a simplified 3-band version of the same parametric EQ), *R* (a cut-down semi-preset version

their sophisticated rackmounting expertise translates to software-only solutions, and (just as importantly) how much processor power these consume.

GOING NATIVE

After installation (and entering the unique serial number for the protection system), you will find two new plug-ins in your DirectX list: *TC Native EQ Graphic*, and *TC Native EQ Parametric*. In addition, if you use either *Cubase VST* or the latest version of *Wavelab* 2.0, you will also find another labelled *TC Native EQ Parametric VST*. This is useful, since DirectX-compatible plug-ins need another layer of code when running inside a VST-compatible application (in fact, given the option, you should always use a VST-specific version, as it will take slightly less processor overhead.)

Both EQ windows have gorgeous 3D graphics, although I'm not convinced that the display windows really need simulated glare! However, the most important part is the user interface, and here TC Works really know what they are doing. Every control is easy and intuitive to use, and you will know 80 percent of what you need within the first few minutes, without even looking at the manual or help file.

EQ GRAPHIC

The EQ Graphic has a level meter at the top left-hand side, with stereo input levels shown above, and outputs beneath. There is a useful peak hold facility (a menu can be called up by right-clicking over the meter), which indicates the highest peak level to date. Beneath the meters are input and output level controls (scaled in linear fashion from -96 to 0dB), and beneath these a dual-function window which normally shows the name of the current ROM preset (a drop-down menu appears when you click on the associated ROM button). However, if you click on the SoftSat button this window changes to a valve-style magic-eye level display. You might remember those on early open reel tape machines: both ends start to illuminate as the effect is introduced, and the display eventually overlaps to give a brighter section in the middle when the level is getting too high. A Compare button also lets you perform before/after comparisons, and right-clicking this brings up an Undo history of the last 20 editing steps.

The right-hand side of the plug-in is dominated by the Touch Screen, which shows level between +12 and -12dB in the vertical direction, and frequency from 20Hz to 20kHz in the horizontal. The equaliser defaults to 28-band operation, but three buttons beneath the display allow you to select between 28-band, 14-band or 7-band options. You can reset the entire response to flat at any time, although this is a permanent change (unless you use the Compare functions), and you should use the application Bypass button for in/out comparisons.

The Touch Screen is really where the fun starts. ▶

The graphic EQ plug-in can be switched between octave (7-band), half-octave (14-band), and third-octave (28-band).

of the full Native Reverb), and *X* (a switchable soft/hard knee compressor).

Many digital EQ designs have a reputation for sounding 'cold', which is why some engineers still prefer to use analogue equalisers. However, along with many other special features, both the *Native EQ* modules (along with the compressor and EQ module of the *Native Essentials* bundle) feature SoftSat, a proprietary algorithm that claims to generate the warm sound associated with analogue equipment. TC have many years' experience of designing hardware effects units (under the name TC Electronic), so it will be interesting to see how

World Radio History

TC WORKS PC PLUG-INS

The parametric EQ plug-in provides a joystick control for tweaking treble and loudness — your PC will thank you if this is the only game you play.

'HELP! I NEED SOMEBODY...'

I do wish plug-in developers could agree on who provides the Help button for them — the plug-in or the host application. The only applications that provided a button to access the help file while running any of these plug-ins were *Sound Forge* and *Cakewalk Pro Audio*. Neither *Wavelab* nor *Cubase VST* had this option, although you can normally locate the help function via the *Windows 95* Taskbar.

▶ The first obvious thing to do is to left-click and drag individual bands up and down. However, much quicker for initial setting-up is to right-click and drag horizontally. This draws a frequency response curve, and once you release the mouse button every individual band jumps to the desired position. To the left there is a vertical slider which provides +/- scaling, so that you can raise or lower every band in the display in proportion. You can also group any combination of bands by holding down the Shift key before clicking on them: changing the level of any band within the group will then affect them all. No processor resources will be used by any band that remains at 0dB.

FEELING PEAKY

The parametric equaliser has a similar setup for the metering and level control as the graphic version, but this time with levels from -96 to +12dB, and with separate SoftSat and ROM preset display windows. The right-hand side of the plug-in has a smaller frequency response display, with the bulk of the controls beneath. There are seven fully adjustable bands, and each of these can be switched in or out of circuit (if you don't need all seven, switching out the unused ones will reduce processor overhead). If you hold down the Shift key, you can globally enable/disable every left or right EQ band by clicking on any left or right button.

A Type button below each in/out switch allows you to select from Parametric, Low Shelf, High Shelf, or Notch characteristics. Beneath this there are readouts of Frequency and Bandwidth, and clicking on either of these brings up a vertical floating fader to make any adjustments that you need. Any of the seven bands can be tuned between 20Hz and 22kHz, and the B/W control either sets the slope in the case of shelving EQ (the options being 3, 6, 9, or 12dB/octave), or the bandwidth (from 0.1 to 4.0 octaves). You can also type in the desired values directly.

The vertical faders provide +/-18dB of range for each band, and are in stereo-linked pairs. You can separate them by clicking on the global Link button — useful, say, for creating spaced pseudo-stereo filtering. If you select a Notch filter response, the faders disappear and the B/W changes to Q, with a different scale of 1.0 to 100. To help you when setting several bands of similar parameters, there are options to copy individual values or the entire band to another one.

The final performance control is a joystick (which has its own in/out switch), and this provides a general treble control in the vertical direction (a shelving EQ of about +/-6dB at 4kHz by the look of it), and the familiar smiley (or frowny) curve loudness response from left to right. Giving the joystick a wiggle gives some very useful tilting responses on top of whatever curve has been set by the parametric, so this is a useful tool to fine-tune an existing EQ.

IN USE

Both equalisers have user interfaces that are easier to get to grips with than other packages, especially if you want to dial in a specific EQ to emulate the response of, say, a particular analogue desk. The fact that any band set to 0dB uses no processor resources is very useful, especially if you only need to make a few changes. Many designs take the same large overhead, no matter how many bands are in use.

TC Works themselves say the SoftSat option is subtle — it's certainly not a replacement for a valve overdrive box or enhancer. But it's nevertheless useful, since it can be difficult to avoid clipping when adding large amounts of boost at a particular frequency. SoftSat generates a soft harmonic distortion, starting at about 4dB or 5dB below clipping according to the main level meters, and stops anything clipping. If you do really overdrive it as an effect, it does make the bass end more punchy, but this is not its main function.

The sound of any EQ is very much a personal thing, but I found both plug-ins powerful and precise, with far more control than the average small desk. By comparison, the sound of the *EQ1* (included with *Cubase VST* and *Wavelab*) was good, but harsh at more extreme settings. More suitable for a direct comparison was the *Q10 Paragraphic* (part of the *Waves Native Power Pack* bundle) and here both EQs acquitted themselves well — the *Q10* has a clean sound with a huge range of options, although it can be tricky to set up. However, I still felt that the *TC Native EQ* had the edge. With the SoftSat switched in, it became warmer at the bass end, and it was certainly a lot easier to use.

I did have a few small niggles, though. When using higher Q levels, the notch filter setting has a tendency to ring, which rather undermines the effect of the preset 50 and 60Hz hum filters. Of course, you always have the option of reducing Q (at the expense of widening the notch). The level faders have a linear law — halfway across they are already 48dB down, so the more useful 0 to -20dB area ends up crammed into the final 20 percent of ▶

TC WORKS PC PLUG-INS

The Native Essentials Reverb is rightly named, giving a quality sound with a significantly lower overhead than its more exotic competitors.

travel. Also, the meters, while graphically attractive, have very small markings.

NATIVE ESSENTIALS

The *Native Essentials* bundle is an altogether less expensive proposition, and at £149 should appeal to many people who want basic effects to add to those already supplied with applications like *Cubase*, *Wavelab*, and *Sound Forge*. There are three entirely separate components on offer: the *R* reverb, *X* dynamics, and *Q* equalisation. Each is an entirely separate plug-in and, like all TC Works products, each is designed to minimise processor usage while maintaining high audio quality.

Some of the features are the same in all three Essential plug-ins — the I/O display is largely the same as both *Native Reverb* and *EQ* plug-ins, showing a stereo input level meter in the top half, and stereo output level meter across the bottom half of the window.

Right-clicking on the meters gives another menu with options for Peak Hold display, but a useful addition (not available in the more expensive ones) is Use VU, which globally deactivates all the metering in order to save several percent of processor overhead. The other main difference is that the Input and Output faders are now incorporated into the I/O display. I noticed that the law has been greatly improved. Although they both still operate over the full 96dB range, most of the useful range is now spread out, so that the top 9dB occupies half of the fader range which makes it far easier to tweak values.

R REVERB

The Reverb plug-in features a simplified version of the *Native Reverb* interface, although there are some differences. Gone are the *Native Reverb*'s more specialised controls to adjust Density, Color and Shape — these are now internally preset. In the central window are displayed the current settings of ROM (18 fixed preset reverb types), Decay Time (from 0.7 to 5.0 seconds), and Mix (0 to 100 percent). Clicking on one of the associated buttons enables you to adjust the appropriate parameter using the large rotary control to the right, or you

can use the mouse buttons to increment or decrement the text values directly.

The Room Size parameter which appeared in the *Native Reverb* (and which scaled early reflection patterns) still appears in the display window, but can only be changed by selecting different presets. Mind you, there are still seven sizes available — from Small through to Huge, along with Spring. The only two of the *Native Reverb* options missing are Tiny and Box, which are no great loss. There are sounds ranging from small Bright Rooms to Huge Cathedrals, with a wide variety of smooth to gritty reflections, and brittle to dark decays. There was no obvious metallic coloration in the longer decays: this is definitely not the poor relation of the *Native Reverb*.

Mind you, processor overhead was significantly lower, and measured 32 percent with my Pentium 166MHz MMX processor (31 percent with Use VU option switched off). This compares with 44 percent for the *Native Reverb*. I tried matching the default settings of the supplied presets to the *Native Reverb*, and got remarkably close, which suggests that much of the same algorithm is present. TC Works subsequently informed me that exactly the same reverb engine is in both, and that only the control options have been restricted in *Essentials*. The reason for the lower overhead is further code optimisation, and by the time you read this, *Native Reverb* owners can expect a free update to the lower-overhead version as well.

Overall, this is very impressive. At an equivalent price of about £50, *Native Essentials*'s Reverb makes the reverbs included with most applications sound like a tin bath, losing out only in terms of flexibility to its more expensive cousin and the Waves *Trueverb*. Given that it takes 30 percent less overhead than the *Native Reverb*, and 40 percent less than *Trueverb*, this is a bargain in more ways than one.

Q EQUALISER

The equaliser is essentially three bands taken from the *Native EQ*. Each can be individually switched in or out, and can be Low Shelf, Parametric, Notch, or High Shelf, with a range of +/-18dB. Three boxes for Frequency, Bandwidth, and Gain, use the same floating fader or text-entry system of the larger version.

A more immediate way to get hands-on control is to use the joystick. Any combination of the three bands can be assigned to it using the three side buttons, to control both frequency (horizontal) and gain (vertical). Initially, control is over the Absolute range, giving frequency from 20Hz to 20kHz, and gain from +18 to -18dB. Although one's first impression is that the corners of the joystick should be 'square', these only restrict the extreme settings, and few people will worry about losing the ability to have +/-18dB swing at 20Hz and 20kHz! However, you can restrict the joystick control range in either or both directions to more manageable proportions.

The sound seems to be exactly that of the *Native EQ*, with similar low overhead. People may ▶

TC WORKS PC PLUG-INS

The Essentials EQ is a versatile 3-band effect that provides shelving and notch options, as well as the more common parametric type.

The X Dynamics plug-in (shown here as an insert in Cubase VST) provides a wide range of sounds, with a choice of soft-knee as well as hard-ratio compression.

> "At an equivalent price of about £50, the *Native Essentials* Reverb makes most free reverbs included with applications sound like a tin bath."

▶ question the need for a general-purpose EQ when most applications provide these as standard, but I do think that this one sounds significantly better than many, and of course SoftSat does add a certain something to the end result.

X DYNAMICS

The final plug-in is a comprehensive compressor, which has traditional controls for Attack (0.1 to 10mS), Release (20mS to two seconds), Threshold (-60 to 0dB), and Ratio (1.0:1 to 64:1). All four controls feature click and draggable rotary knobs, although as usual you can type chosen values in directly as well. The amount of gain reduction is shown in a small meter on the right, and beneath this are two switches, one to toggle between soft- and hard-knee compression, and the other to activate SoftSat.

There are a dozen presets supplied, and these range from subtle to extreme, though all are effective. At this price point, having the soft-knee option is very welcome for providing less processed sounds, and the SoftSat option seems more pronounced than in the other modules, giving an even wider range of options.

SUMMARY

The TC Works range continues to provide high-quality DirectX plug-ins. The *Native Essentials* may not be essential to everyone, but it is a carefully thought-out and useful bundle. The reverb is truly excellent for the price, and the compressor gives a wide range of options from general correction right through to manic mangling. The EQ also

sounds good — it's versatile, too — but will probably not be quite so high up most people's wish lists. However, as a package, only the *Waves EasyWaves* bundle at £125 comes anywhere near this quality for the price. For the extra £25, *Native Essentials* provides a lot more options and wide-ranging sounds.

If you want a cheap but versatile EQ to add to your audio applications then the one in the *Native Essentials* bundle will be ideal. However, if you want the ultimate in EQ flexibility, along with a very clean sound (or the option of some benign second-harmonic distortion courtesy of SoftSat), the *Native EQ* package should provide everything you need. Both graphic and parametric plug-ins are also so easy to set up that you won't be waiting for a library to be released to take full advantage of them. There is always a place for a toolbox EQ that can do everything. With the *Native EQ* pack, TC have given us the works. **SOS**

Taking the Bits

DIGITALLY TRANSFERRING SOUNDS FROM AUDIO CD TO YOUR PC

To get the maximum quality from audio sample CDs, you need to transfer the sound data to your PC in the digital domain. MARTIN WALKER explains why the process isn't always as easy as you might expect.

Sample CDs now come in three main formats. The first are those that are read directly by a specific make of sampler, such as Akai, Roland or Emu (although many samplers can now read formats other than their own). For computer use, CD-ROMs are also available with sounds in PC WAV or Mac AIFF format. The third type is the audio sample CD, and while these lack the looping and performance information of the previous types, they offer a much wider variety of sounds, at often a third of the price of the sampler-specific CDs. To extract the highest quality from audio sample CDs it's always best to go for a bit-by-bit digital grab, rather than re-sampling the sounds through another A-D converter.

You might also want to do a digital grab in some circumstances if you have a CD-R drive that you use for creating master CDs of your own music for duplication. Once you've burnt your latest album, you might expect that reading back the same audio data into your hard drive would be as easy as the writing process. However, this is not always the case, and the digital audio extraction can cause various unexpected problems.
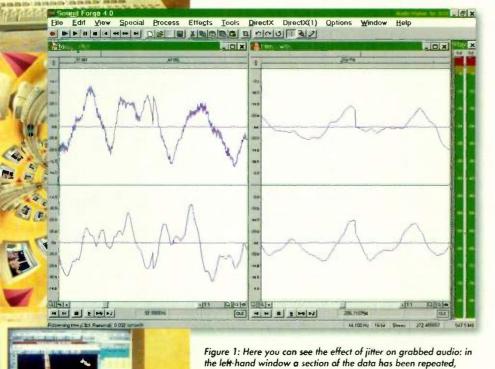
GETTING ON TRACK

According to Adaptec, who market *Easy CD Creator* for PC, and *Toast* (and *Jam*) for Macs, all CD-R drives can perform digital audio extraction. However, many CD-ROM drives do not support it at all, and those that do can rarely perform it reliably at anywhere near their top speed. For instance, the typical 24-speed CD-ROM drives currently supplied with many machines may not manage to extract audio at more than 8x normal speed, and in some cases the only recourse for reliable reading may be to resort to reading at single speed — or, in other words, real time.

Although some SCSI CD-ROM drives can cause problems with digital audio extraction, the EIDE drives used in many PCs are in an even greyer area, since few professional applications even attempt to read audio from them. Although there are various shareware utilities that claim to do this, even here the success rate varies widely, not only between drives from different manufacturers, but also from model to model, and sometimes even depending on the firmware version (operating system stored on a chip inside the drive) that you have.

The reason for these problems is another version of our old friend jitter. Normally, when we refer to jitter we're talking about minute inconsistencies in the timing of digital audio, as it is 'clocked out' by the D-A (digital to analogue) converter. These timing variations can be either periodic (regular), in which case they result in spurious fixed-frequency distortions at low audio levels, or random, which causes an increase in the level of background noise. In the case of digital audio extraction jitter, the mechanism and effects are rather different, and rather more audible if not dealt with.

When an audio CD is being read, the information is usually simply passed into a buffer which acts rather like a bucket containing water. As long as there is water (data) in the bucket, a highly accurate clock reads it out at an extremely steady rate (the accuracy of this determines the amount of traditional jitter). The spindle speed of the CD player (how much water is being trickled in) is simply adjusted to make sure that there is always water in the bucket. However, for digital audio extraction, the computer system has to grab a ▶

Figure 1: Here you can see the effect of jitter on grabbed audio: in the left-hand window a section of the data has been repeated, whereas in the right-hand window the vertical jump in both channels indicates a section that has been missed altogether.

World Radio History

DIGITAL TRANSFER FROM AUDIO CD TO PC

Easy CD Creator Options

Figure 2: The latest 3.01 version of Easy CD Creator has audio re-syncing options, which should make glitches a thing of the past with most drives.

▶ chunk, write this data to the hard drive, and then return to exactly the same point in the data stream. Since the absolute position in the data stream of the previous grabbed block is now slightly uncertain, the computer may not always continue extracting at exactly the same place in the stream of audio data that it left off. The extraction process may thus continue from a slightly different place in the stream. If it restarts from a point a few bytes earlier in the stream, part of the waveform will get repeated; a few bytes later, and there will be a gap in the waveform. Either way there will be a sudden discontinuity in the grabbed waveform, which will result in an audible click on playback. (See Figure 1.)

The difference with a CD-ROM is that each block of data contains a header, with sync information, and a copy of the block's address, so that the drive can easily find the start and address of each block

— vital for data storage. When it comes to CD Audio, apparently only Plextor CD-ROM drives have special extra hardware built in to guarantee the absolute position of audio data, although I have heard some conflicting reports about this.

Some drives have problems if the pre-gap before the first audio track isn't exactly two seconds (the expected default timing). A few drives can't cope with this, and start digital audio extraction slightly late, so that they also read slightly beyond the end of the track. Fortunately this problem only seems to be associated with a some older drives, such as the Yamaha CDR100/102 and the Philips CDD2600, but if you're creating CD-Rs of your own music it would seem safest to leave the initial two-second gap alone, whatever you do with timing between the other tracks.

TRYING IT OUT

Many drives that support digital audio extraction may not achieve reliable results at their higher speeds, so some experimentation is in order. If you try grabbing an audio track with your CD-ROM drive operating at its maximum digital audio extraction speed, you may hear obvious problems as soon as you listen to the result, or you may just hear the odd tiny tick or pop in the grabbed file. An occasional glitch may not worry you, but if you

> "To extract the highest quality from audio sample CDs it's always best to go for a bit-by-bit digital grab, rather than re-sampling the sounds through another A-D converter."

want to guarantee that a read will work reliably every time, slow down the grabbing rate, so that your system has more time to ensure proper synchronisation of the audio data. This involves a bit of trial and error, but you'll probably only need to try three or four times to establish the maximum reliable grabbing speed of your drive.

A particularly bad drive may only extract at single speed, and in this case it will be impossible to back up your audio CD in real time to another CD-R drive, since even if you reduce your CD-R to 1x write speed it's unlikely that the audio extraction will be able to reliably keep up with the writing process. ▶

COPY RIGHTS?

Being able to digitally copy audio CDs opens up the tricky question of copyright. In this country, people have been making copies of their own CD or vinyl albums onto compact cassette for many years, but the whole issue remains a hazy one. The new consumer CD-R recorders (such as the Philips CDR870 reviewed in the

December 1997 issue) can only use special consumer CD-R blanks, and the price of these includes an amount which goes towards any copyright fees that may be lost.

However, even huge companies like Adaptec are now marketing programs such as Spin Doctor, which are designed to help you transfer your vinyl collection to CD-R, complete with pop and crackle reduction.

They do include a red sheet inside the box, advising you to contact your legal adviser before copying anything.

The bottom line is that is that we musicians can see both sides of the fence. If you do have a CD-R drive, and are using it to compile your own music, think on before you copy someone else's — one day you may be relying on sales of your own CDs to earn a living.

Figure 3: If you've ever wondered why some CDs are fatiguing to listen to, grabbing a track and examining it in a WAV editor can prove instructive. This one could be described as powerful or lacking in dynamics, depending on your viewpoint.

► You'd then have to create an image file of the source CD on your hard drive, and write from that. This is usually the most reliable method anyway, since there can be various problems when performing digital audio extraction while using the same buss (SCSI or otherwise) to write data. Anyone who read the *SOS* review of the Yamaha CDR400tx (in the December 1997 issue) may remember that I suffered from low-level 'ticks' in a CD-R that I burned while simultaneously using digital audio extraction to read an audio CD using a TEAC CD-ROM drive. I now know what caused the problem.
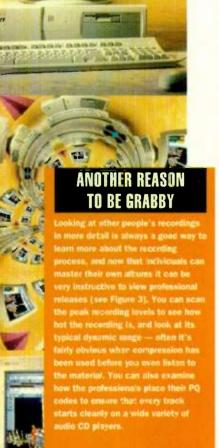
One software solution to this hardware problem has been adopted by various pieces of utility software. It's called re-syncing (or jitter correction), and relies on grabbing larger, overlapping sections of audio. These sections are then moved backwards or forwards until the overlapping sections match exactly, at which point they will be exactly in sync. This technique is used in the latest version of Adaptec's *Easy CD Creator* (see Figure 2). The procedure with this program is first to run the system tests, to determine the maximum extraction rate for your drive. You can choose from three selectable speeds for digital audio extraction: the highest (default) setting simply uses the maximum speed already established by these tests. The medium speed still allows CD-R drives, and a few specific well-behaved CD-ROM drives, to extract at the maximum speed, but uses re-syncing for all other drives. This is a sensible setting if you have a CD-R and a less reliable CD-ROM in the same machine. If you experience any glitches, switching to the Slow speed always uses audio re-syncing, but this significantly reduces audio extraction speed. You may find that personally establishing a maximum reliable extraction speed for your drive will achieve faster results than resorting to audio re-syncing.

TESTING YOUR DRIVE

The only absolute test of accuracy is to grab a track several times at a certain read speed, then compare the resulting files. If there are no obvious audio glitches or ticks you should be safe to continue grabbing at this speed. Those with noise-reduction plug-ins can also use any 'find click' functions, which will speed up the search. If you find anomalies, reduce the CD read speed in your grabbing software and try again.

Another way to double-check the data is to use the DOS utility FC.EXE (File Compare) that comes with Windows 95. You'll need to open a DOS window to do this, navigate to the folder containing your WAV files, and then type the command line 'fc/b File1.wav File2.wav' (with File1.wav and File2.wav being replaced by the names of your own two files). The /b part ensures that a byte for byte comparison is carried out. Any bytes that differ will have their respective values printed out on screen. By the way, if you do get stuck in a seemingly endless printout of different bytes, you can use the Ctrl+C or Ctrl+Break key combinations to escape.

Using *Wavelab* 1.6, my attempts to grab at Auto (fastest) speed with the Yamaha CRW4260 at 6x read speed produced occasional glitches, but at 4x speed I didn't hear a single one. However, when I'd grabbed a two-minute track six times, using FC.EXE suggested that every single byte was different every time but one, since only one pair of the six files proved to be identical. I have seen references on the Internet to this non-repeatability of data, but although the numbers are different, the audio always seems to sound the same. This didn't make sense to me at first, since it suggested that the data was being modified in some way, so I investigated further. The big clue came when I examined the actual bytes in a binary

file editor. After some zero bytes (digital silence), the first few non-zero bytes of each of my six files were identical, but not their position within each file; this varied by anything up to 16 bytes.

Since *Cool Edit Pro* allows you to view data right down to byte level, I used the program to carefully trim off all the zero byte values from the beginning of each file, so that each waveform started at precisely the same place. After this, all six grabs proved identical when tested with FC.EXE. So the answer is that the timing uncertainty will normally give you identical audio files, but with slightly varying numbers of zero bytes at the beginning. This is perfectly acceptable, and seems to prove that you can always get back exactly what you burned onto an audio CD-R, though I can't guarantee that this will be the case with every drive.

A BIT TRICKY

If you have yet to buy a CD-ROM or CD-R drive, the possibility of audio extraction will probably enter into your choice. However, attempting to check this aspect of technical performance can be a tricky business, since manufacturers' specifications often tend not to mention digital audio extraction at all. Buying such drives is a gamble, since although many modern drives provide this facility it's still by no means universal, particularly for PC EIDE types. You are also unlikely

to see this feature quoted in advertising, although some specialist music dealers may know whether the models they stock can extract digital audio.

Often the only place to turn is the web site of a particular manufacturer, where technical details are normally available in full. You can sometimes even download a complete copy of the user guide. However, even if the drive supports digital audio extraction it's often difficult to find out the likely maximum speed without a test run. Drives generally recommended are from Plextor (all models), the Pioneer 10x speed CD-ROM, and Sony.

As I said earlier, from my own experience it's always safest to create an image file on your hard drive before attempting to burn a CD-R disc, to minimise the chances of glitches. Don't forget to defragment your drive before you start the process: the last thing you want is to make the drive's head assembly jump about more than is needed. The other advantage of creating an image file is that it gives you a chance to check the audio before you waste any CD-R blanks on duplicating glitches.

If you're compiling your own music for an album, I suggest that you burn an additional data CD-ROM of the separate tracks in WAV or AIFF format for backup purposes, and not rely totally on an audio CD version for posterity, since this provides added error correction. Aren't computers wonderful?   **SOS**

"Many drives that support digital audio extraction may not achieve reliable results at their higher speeds, so some experimentation is in order."

NAMM show at the start of the year, the complete Evolution Series comprises 10 models, six of which are dedicated vocal mics, the E825s, E835, E835s, E845, E845s and E855. An "s" suffix simply means the model has a switch, so only the E825s, E835s, E845s and E855 are reviewed here. For recording and miking instruments, there are also the E602, E604, E608 and E609 models, which we'll be checking out in a separate review.

Despite the fact that the lowest model in the range is extremely inexpensive, all the mics are finished to the same high standard of engineering and feature a heavy, blue/grey metallic-painted body combined with a very tough wire basket that can be unscrewed for cleaning. Virtually every mic manufacturer claims to have done something special in the way of capsule damping to minimise handling noise and Sennheiser are no exception. The handling noise is acceptably low for live use but, as ever, hand-held use is not recommended in the studio as some low-frequency noise still gets through. Each of the vocal mics comes in a presentation box complete with soft plastic zip-up pouch and a stand clip, but the cable is not supplied. All the models in the Evolution series have a low impedance balanced output on a conventionally wired XLR connector.

ORIGIN OF A SPECIES

The fact that most vocal microphones now look very much alike is a result of evolution — a tapered handle with a ball-shaped wire grille on top is easy to handle, it looks OK and it protects the capsule from physical damage as well as affording some protection against breath blasts that cause popping. Occasionally, mic designers try to break away from the 'ball on a stick' norm by using an egg-shaped basket or by flattening the the top of the basket, and something of the sort has been tried here. While the baby of the series, the E825(s) has a traditional appearance, the remaining three vocal mics have flat topped baskets.

evolution now!

SENNHEISER EVOLUTION SERIES VOCAL MICROPHONES

PAUL WHITE studio-tests Sennheiser's new range of affordable vocal microphones to see if they really can give the established models a run for their money.

Sennheiser are one of the big names in microphone manufacture, but like all the other big names, they've chosen to launch a budget range in order to capture a larger share of the cost-conscious gigging musician and home studio market. Launched at the Los Angeles

The cardioid E825s is unashamedly an entry-level cardioid vocal microphone, but you wouldn't guess its price from looking at it, or from listening to it for that matter. It has the smooth lines and impeccable finish of a top-quality European microphone. Of the Evolution vocal mics, it has the most restricted frequency response, specified as 80Hz to 15kHz, augmented by a broad, gentle presence peak centred around 6kHz. This lends the sound a slightly middly quality, but it's not excessively nasal or unnatural and actually cuts through extremely well in a crowded mix. The

switch operates smoothly and is positioned so as to make unintentional operation unlikely.

The cardioid E 835s has a similar shaped body, but this time the basket has a flat top. The frequency response is rather wider than that of the E825s, at a quoted 40Hz to 16kHz, but it still has a solid sound that projects well. It's also electrically more efficient than the E825s, producing a noticeably higher output for the same input. As with most vocal mics, there is a presence peak, though it's actually quite subtle sounding, so once again the result is fairly natural.

Unlike the previous two models, the E845s has a supercardioid polar pattern. This is designed to help reduce spill and feedback under live performance conditions, but may also help in the studio when you have two or more players working at the same time and you want to keep the spill level down. As with all hypercardioid mics, there's a rear-pointing lobe that makes it more sensitive to sounds coming at it directly from behind than a regular cardioid, so you need to be aware of where the rear of the mic is pointing. Again the response is 40Hz to 16kHz, and this time the presence boost is between 4kHz and 5kHz and is fairly gentle, which avoids the nasal honkiness some vocal mics produce. The overall tonality isn't much different to that of the E835s, and if I had to compare it with the ubiquitous Shure SM58, I'd say it was slightly more open and brighter sounding, but otherwise not dissimilar.

Also featuring the flattened-basket styling of the previous two models, the supercardioid E855 is the top vocal mic of the range. Its frequency response extends from 60Hz to 18kHz, and a slightly modified presence peak gives the mic a little more clarity under difficult conditions, but without sacrificing warmth and depth. In deference to the superiority of this mic, it comes without a switch.

PERFORMANCE

As you might expect, the performance improves as you move up the range, but the entry level E825 works a lot better than you'd imagine it would for the price. It's a little less sensitive than the other mics in the range, but it actually sounds quite natural, and once you get close up the proximity effect provides the necessary warmth. This would be a suitable mic for vocal recording on budget multitrackers and similar systems, as well as being an acceptably good live vocal mic, though it also works well on electric instruments such as guitar amps.

As soon as you move up to the E835, the sound warms up a little and the efficiency improves, while at the same time the middly character is reduced. It's always hard to describe the way a mic sounds, but I did direct comparisons with a number of other dynamic models, and found that the overall character came close to that of my ageing SM58, but with slightly less low end warmth and better high end detail.

The two supercardioid models performed extremely well, which isn't surprising considering

they're at the more expensive end of the price scale, but there's generally little benefit in using a supercardioid for vocal work in the studio as the timbre can change as the singer moves relative to the mic while performing. Even so, the E855 sounded exceptionally nice, and as with all the mics in the range, it doubles quite happily as an instrument mic.
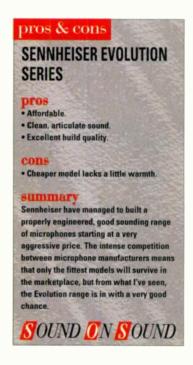
SUMMARY

The dynamic microphone market is hugely competitive, and there's also a lot of inertia that tends to push people towards buying tried and trusted models. Occasionally, however, a new range comes along that pushes the boundaries a bit further. Being honest, I don't think that, measured by performance alone, the Evolution range really does offer anything new, but what it does achieve is a remarkable balance of build quality and performance at a very attractive price. Most dynamic vocal mics are designed with the live market in mind, and these are no exception, but they all work fine for recording, even the budget E825. If you're looking for quality dynamic mics but don't have a big budget, then the Evolution range should meet your needs and exceed your expectations. [SOS]

World Radio History

World Radio History

World Radio History

Music Village

Masters of Sub Bass, Breaks and Beats

10 High Road, Chadwell Heath, Essex · 0181 598 9506   86 Mill Road, Cambridge · 01223 316091

562 Brighton Road, South Croydon · 0181 407 8444   230 High Street, Barnet · 0181 440 3440

THE SAMPLING & SYNTH CENTRE OF THE UNIVERSE

If you take your sampling seriously there's only one place to call – all our sampler packages come with FREE unlimited access to the famous Digital Village sample library!

As the spiritual home of Akai and Emu sampling, we can offer you the very best prices.
If someone else tempts you with an offer – pick up the phone – <u>Talk to the Deal Makers</u>

Emu E6400
Super Deal £1799.99

FREE DIGITAL VILLAGE LIBRARY ACCESS

including Planet Phatt and Orbit Rom RRP £399 or Soundcraft Folio 10/2 mixing desk with swept mid or FREE CD-Rom drive and 16 Meg.
All E6400's from the Village come with 5 FREE CD-Roms.

E6400 + Waldorf Pulse	£1799.99
E6400 + TLA Ivory 5021	£1899.99
E6400 + Supernova	£2599.99

ESI-4000

FREE DIGITAL VILLAGE LIBRARY ACCESS

- 18 Filter types   • 64 Voices
- Expandable to 128 Meg
- Up to 10 outputs

32 Meg + Zip drive + CD-Rom drive + 5 **FREE** CD Roms

ESI4000 + Calamari Turbo	£1199.99	**VILLAGE EXCLUSIVE** £1199.99
ESI4000 + QY70	£1299.99	
ESI4000 + Supernova	£1999.99	
ESI4000 + Waldorf Pulse + AKG 270s Headphones £1199.99		

E-Synth & E-Synth Dance

FREE DIGITAL VILLAGE LIBRARY ACCESS

- Up to 32 Meg of onboard waveforms (16 Meg Orbit & Planet Phatt waveforms on the dance model)
- Expandable to 128 Meg   • Up to 16 outputs
- Includes 5 **FREE** CD-ROMs

Ex-demo ESI-32 (3 left) £549.99

GUARANTEED BEST PRICE

Planet Phatt and Orbit ROM
All the sounds of these classic EMU modules on one 16 meg Rom board – The ultimate dance upgrade for the E6400, E-Synth, E4X £399.99

ASK FOR SPECIAL DEALS ON THE DANCE ROM WHEN YOU BUY AN EMU E SAMPLER

Emu Audity 2000
Digital Modular Rhythmic Synthesizer (DMRS) £1299.99
- 640 presets   • 16 part multitimbral
- 4 layers per preset

Emu Orbit
The Ambient, Trance and Techno
TOTAL PRODUCTION TOOL

GUARANTEED BEST PRICE

Emu Planet Phatt
The Hip Hop, Trip Hop, Swing, R & B, 70's Groove machine

GUARANTEED BEST PRICE

External CD-ROM drives from only £99.99 Zips & EZ Flyers always available

Akai S5000
- 3U • 8 Outputs (exp to 16)
- 64 note poly (exp to 128)
- 15 Filter types as standard
- EB20 FX can be fitted

FREE DIGITAL VILLAGE LIBRARY ACCESS

ONLY £1799.99

Akai S6000
- Removable front panel/remote controller
- 16 Outputs as standard   • 128 note poly
- 15 Filter types as standard   • 4U
- EB20 FX as standard

FREE DIGITAL VILLAGE LIBRARY ACCESS

ONLY £2799.99

We have a trailer load coming late summer / early autumn.
Phone now to reserve yours – they'll go fast!!

Akai S3000XL
- Expandable to 32 Meg
- 10 Outputs
- Industry Standard

FREE DIGITAL VILLAGE LIBRARY ACCESS

VILLAGE PRICE Including 32Meg £1099.99

Akai MPC2000
- Sequencing / Sampling Drum machine
- SCSI • Waveform editing
- Rock solid timing

FREE DIGITAL VILLAGE LIBRARY ACCESS

VILLAGE PRICE including 32Meg £899.99

Akai S2000
- Expandable up to 10 outputs   • Same spec & quality as S3000XL • Including 16 Meg for only £649.99

Akai S20 … £399.99   CD3000XL … £1099.99

Zoom 234
Drum Machine £199.99

Crunchy beats and low basses

Mutator Stereo Filter	£529.99
Mutator + Midi	£589.99

Voyager Beat Xtractor
SLAVE TO THE RHYTHM
- Audio BPM to midi interface
- Syncs record decks to sequencers

£249.99
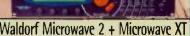
Sherman Filterbank	£499.99
ATC1 (1 left)	£439.99

Waldorf Pulse and Pulse+
- 3 Oscillators • Arpeggiator • Matrix editing
- MIDI syncable • LFO, CV gate in/out
- External processing on Pulse+

prices from £279.99

Novation Supernova
- 16 Note poly
- 8 Part
- FX on every part

£1299.99

BassStation & DrumStation – Guaranteed Best Prices

Waldorf Microwave 2 + Microwave XT
The Microwave moves on
- 10 Note poly • Ring modulation • Arpeggiator
- Rich resonant sweeps and gut wrenching subs

prices from £749.99

Nord Nord Lead
Modular rack and keyboard and entire Nord range always in stock and on demo.

prices from £995.99

World Radio History

World Radio History

AUDIO ALCHEMY?

APOGEE AD8000 24-BIT DIGITAL AUDIO CONVERTER

The AD8000 is a high performance 8-channel, 24-bit A-D converter which can cope with an array of different digital audio formats, either in its basic form or with the addition of one of several optional cards. MIKE COLLINS gets converted...
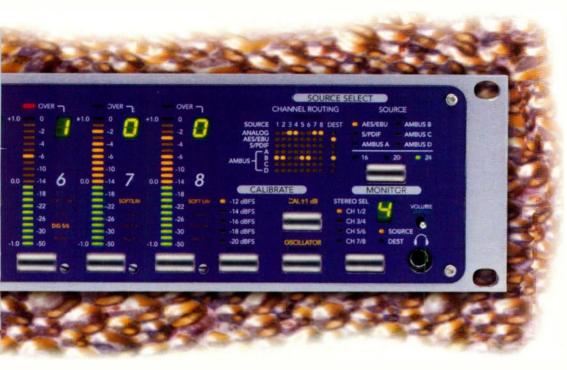
pogee's new AD8000 is an 8-channel, 24-bit A-D converter and processor which offers very low noise figures (-110dB THD + Noise), has a 114dB dynamic range, and a frequency response that's flat from 20Hz to 20kHz, plus or minus 0.025 dB. The basic unit has four stereo AES/EBU outputs and one stereo AES/EBU input, plus stereo S/PDIF input and output — and an Apogee Multimedia Bus (AMBus) card is available, for a further eight channels of AES/EBU input. You can install up to four of the optional AMBus cards to provide interfaces for Pro Tools, ADAT or TDIF. In practice, this means that the unit can be used as a format converter to convert from ADAT to TDIF to Pro Tools to AES/EBU — or any combination of these. The ADAT and TDIF AMBus cards also have built-in Apogee Bit-Splitting, to allow recording of six channels at 20-bit resolution or four channels at 24-bit resolution on to eight 16-bit digital multitrack tape tracks — for example, on to a Tascam DA88 or Alesis ADAT. This makes the AD8000 ideal for preparing audio masters for the new 5.1 surround and 20- or 24-bit audio formats.

Other options include a stereo 24-bit DAC card with dual XLR outputs, an 8-channel 24-bit DAC card, and a remote microphone pre-amp controlled from the AD8000. An optional Sample Rate Converter will also be available later this year. This will install in one of the AMBus slots and will allow any sample rate between 32kHz and 54kHz to be fed in and output at 44.1 or 48kHz, via any selected output, whether AMBus or AES. A video sync board is yet another AD8000 option.

FEATURES & CONNECTIONS

The AD8000 features a low-jitter master clock, and 'Smart' sync capabilities are provided to automatically sync to word clock, the optional video board, or any digital input. Switchable analogue and digital DC offset removal is provided, and connections are on standard XLRs for analogue I/O and RCA/phono jacks for S/PDIF inputs and outputs. The unit also boasts built-in UV22 processing, which allows you to convert from 24-bit down to 16-bit while preserving low-level information. Other features include a built-in test-tone oscillator and a headphone monitor for any two of the eight channels. This latter feature, which isn't available on Digidesign interface units, was very useful when I was setting up and troubleshooting the system.

The range of connectors on the AD8000 should give you some idea of its flexibility. The back panel offers eight XLRs for balanced analogue line input, with four XLRs carrying four pairs of AES/EBU digital outputs, and one XLR for stereo AES/EBU digital input. There's also a pair of RCA/phono co-axial connectors for S/PDIF in and out, a BNC for word clock or video sync input, and another for word clock out, plus a 9-pin comms port and a block of ten DIP switches. These DIP switches let you set various parameters, such as how many consecutive full-scale samples constitute a digital overload, or whether to use +4 dBu or -10 dBV line levels, and you can also enable the DC removal feature for the analogue or digital inputs. If any DC (Direct Current, as opposed to the Alternating Current of the audio)

is present on incoming signals, this would normally be passed through to the output signals. If it became necessary to edit this later on, any section of audio without a DC component edited on to one with a DC component would cause a loud click at the edit point.

FRONT PANEL

The front panel is where most of the action is. At the far left is a Power on/off switch, and buttons for Sync Source and Sample Rate are situated immediately to the right of this. The Sync Source button selects between the internal Crystal oscillator and external sync sources, including the word clock, AES/EBU and S/PDIF inputs on the back panel. It's also possible to lock to any device which has an active output connected to an AMBus card installed in one of the slots. With the optional Video Sync module installed you also get the option to sync to PAL, NTSC or Monochrome external video sync signals, which are automatically detected and locked to. In this case the AD8000 can also do 'pull-ups' and 'pull-downs' to compensate for the speed changes necessary when transferring NTSC video to and from film. For this you can choose to multiply or divide the sample rate by 1.001 using the sample rate button, which also allows you to choose between 44.1 and 48kHz sample rates.

To the right of these controls are three buttons for controlling the meters:

- The **Clear Meters** button lets you reset the displays to zero.
- The **Overs** button lets you choose whether to hold the overs indication, auto-clear the overs, or defeat this function altogether. Choosing Hold or Autoclear causes the 7-segment 'over' counter displays to keep track of the number of digital overs on each channel, up to a total of nine. More than this number of overs is indicated as a '-'. Selecting Autoclear causes the displays to hold the last indication, although they will

automatically clear when any audio above -50dB is detected after 15 seconds of silence.

- The **Meters** button presents a choice of six metering modes, with various combinations of Peak and Average indication, including two-second and infinite Peak Hold The permutations would be rather too long-winded to write down, but the basic message is that the meters are pretty versatile.

To the right of the meter controls there are four more buttons, as follows:

- **Soft Limit:** This helps to get more level onto tape without 'overs', and is the same system used on Apogee's popular AD1000 A-D converters. It can be applied individually to any analogue input channel, or switched on or off for all channels.
- **UV22:** UV22 encoding can be applied to any analogue input channel or any pair of digital inputs, using the UV22 button in conjunction with the channel buttons. UV22 encodes high-resolution 24- or 20-bit digital audio into 16- or 20-bit formats, preserving the audio quality and avoiding truncation. If you're recording to a 20-bit or 16-bit system, you should use UV22 on all channels. If you know you'll be using UV22 at some later stage in your project — when mixing or mastering, for example — you can set UV22 to a 'low' setting using one of the DIP switches on the back panel. Another DIP switch lets you select the appropriate UV22 setting for either 20-bit or 16-bit output.

"Project studios running Pro Tools 24 with an 02R mixer should strongly consider buying one AD8000 and the new Digidesign ADAT interface."

pros & cons

APOGEE AD8000 £4524

pros
- High-quality A-D and D-A conversion for popular digital audio workstations and digital multitracks.
- Extremely flexible interfacing options.
- Also acts as a digital format converter.

cons
- Setting up to use the wide range of features and options can be confusing at times.
- Manual needs to be improved to make the unit more accessible to typical users.
- A bit pricey in absolute terms, although still cheaper than alternatives from other manufacturers.

summary
If you're looking for a better set of converters for your Pro Tools system, ADAT, Tascam DA-format recorder, Yamaha 02R, or whatever, this unit from Apogee is just what the doctor ordered — you get more interfacing options than you can shake a stick at and it looks great as well!

SOUND ON SOUND

►

APOGEE AD8000

One of the AD8000's meters in close up.

▶ • **Digital In:** This button lets you switch between analogue and digital inputs on each channel.
• **Mute:** mutes any channel. The mute function will send 'digital black' (no sound, with clock) to the selected channels.

The most striking feature of the front panel, of course, is the central section, which is taken up by the eight meters, each of which features 21 LED segments ranging from 0dB to -50dB at the bottom of the range. Above each meter there's a single LED to indicate overs, and to the right at the top of each meter there is an associated 7-segment display which shows the number of overs, as explained earlier. If you've selected Soft Limit, UV22, Digital In or Mute for any channel, one of the four legends associated with each channel will light up to indicate this status. It's a very sexy interface, in all, with plenty of visual feedback.

To the right of the main display area there are three more groups of controls: Source Select, Calibrate and Monitor. Let's take a look at calibration first. This is a pretty straightforward procedure. You set up a 1kHz tone on your console's output busses at exactly 0VU on its meters, press the 'Cal' button on the AD8000, then choose a headroom setting. Typically, this would be -16 for digital multitrack, but if you're using Soft Limit you can go to -14 or even -12 if you want to get more level on tape. The 7-segment displays will now indicate by how many dB you are above or below the selected reference level — a '-' symbol indicates a level 10dB or more above or below. A level that's above is indicated by the 0dBFS LED lighting, while one that's below is indicated by the -50dBFS LED lighting. To set the input gain levels, adjust the trim pot on the selected channel until the number indicated in the 7-segment display reaches 0. At this point the entire range of the meter becomes 2dB, with each segment representing a tenth of a dB. In this mode you tweak each channel so that the middle LED is lit. This allows you to calibrate each channel to within 0.1dB of the selected reference level.

The AD8000 also has a built-in digital oscillator which you can select by holding the Oscillator button down for at least one second. This generates a sine wave

at 1kHz across all eight digital outputs (including all AMBus cards) at the selected headroom setting. The oscillator level can be set to any one of the five different reference levels by repeatedly pressing the Headroom Select button until the setting you want is displayed. This is useful for quickly checking connections and routing, and also for recording an accurate alignment tone at the head of your project tape, to use for setting levels.

The Source Select section at the top right of the front panel features a channel routing matrix display (8x7 LEDs), with the Source button and associated LEDs to indicate the chosen source to the right of this. The matrix displays which sources are feeding each of the eight channels, and this is pretty easy to understand and use: the rows represent the sources and the columns represent the output channel numbers. If the top row is all lit, for example, this indicates that A-D conversion is taking place from the analogue sources on all channels through the unit. If you then wanted to transfer from a digital source, you'd simply press the Source button, then the Digital In button, and select whichever channels you'd like to transfer from the digital source.

There is also a column of LEDs marked 'Dest' in this area, which indicates the AMBus destinations for the channels.

MONITOR CONTROLS
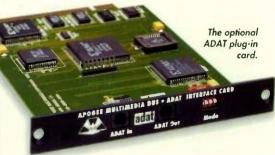
The bottom right of the front panel hosts the Monitor controls. The AD8000 has an 18-bit D-A converter for monitoring. Briefly pressing the Monitor Select button routes audio from the selected pair of channels or the AES input to three destinations: the S/PDIF output on the back of the unit, the D-A converter for the front-panel headphone output, or the optional 24-bit stereo D-A card, if you have one installed. You can't mix tracks together, however. There's a handy little

The optional ADAT plug-in card.

volume control, which only affects headphone volume, next to the headphone output, and an associated level display.

The Monitor Select button is used to step through the input selections — 1/2, 3/4, 5/6 and 7/8 — and LEDs will light to indicate the channel pair selected. Pressing the button once more when 7/8 is displayed lights all four LEDs to indicate that the onboard AES input is being fed directly to the headphone D-A and the optional 2-channel D-A card, if installed.   ▶

AMBUS CARD OPTIONS

A TDIF AMBus card is available for interfacing to Tascam machines, and an AES Input AMBus card adds an additional eight channels of AES/EBU inputs. For recording and overdubbing in mono or stereo you won't need this, but if you want to record digitally onto eight tracks at once, via AES/EBU, you will. Apogee have also developed a fibre-optic system so that a pair of AD8000s can be networked over distances of up to three and half miles using their new Fiber DX AMBus card (just how big are American studios?).

The demo unit came fitted with both DIGI 8 and ADAT AMBus cards. You can use the DIGI 8 card to interface to Pro Tools, as an alternative to using the Digidesign 888l24, 888 or 882 interfaces and the ADAT card interfaces with any ADAT optical

interface. The 20-bit ADAT-format machines are supported, but what if you only have a 16-bit ADAT and you want to record at 20 or 24-bit resolution? Well, Apogee have thought about this as well. Both the ADAT and the TDIF cards offer Apogee's Bit-Splitting technology, as mentioned in the main body of the review, to let you record six channels at 20-bit resolution or four channels at 24-bit, spreading the data over the eight available 16-bit tape tracks. For each 24-bit data stream, 16 bits of data are recorded onto the first of a pair of tape tracks, with the last eight bits being recorded onto the second track of the pair. On replay, these are recombined to form one 24-bit signal. Two AMBus cards with two ADATs or DA88s would allow recording of up to four stereo pairs simultaneously. And, just in case you were wondering, the 20-bit mode works similarly and is primarily intended for surround sound mixes.

"Why has this instrument generated such a buzz? Well, I think it's fair to say that it's a landmark in synthesis."

Paul Nagle, Sound On Sound, April 1998.

An anecdote about Patch synthesisers.

Remember old analog modular patch synthesisers when you would make up one patch after hours of tedious patching? When you created a sound to your taste, ohh it sounded so good, but you needed to totally re-arrange your patch cords and knob settings to create new sounds. The process would have to start all over again and that great mono sound might never be heard again (unless you took the time to draw a comprehensive diagram of your settings and patches). To sample a patch is not the solution. Sampling is static and cannot reproduce random nuances.

nord
MODULAR

Keyboard or rack versions with PC software

- Virtually unlimited oscillators
- Almost unlimited module combinations so you can create almost any kind of synthesiser you want
- 'Virtual Cables' for connecting over 70 modules including: OSC, VCF, VCA, LFO's with more to come.
- Distortion & Overdrive modules for those Hard & Cutting' sounds
- Four Modular synthesisers in one
- 8 Voice Double Capacity Expansion Card
- Real Time Editing
- 100 Memory locations so you can take the keyboard/rack with you to the gig.
- Morphing possibilities in a startling new way

For more information contact Trish on:
Tel: 01245 344001 Fax: 01245 344002
email: info@keyaudio.co.uk

Key audio
systems

APOGEE AD8000

The AD8000 can be put into what Apogee refer to as Destination mode if the Monitor button is depressed for at least one second. This mode is used during recording for overdubbing and confidence monitoring through any recording device connected to an AMBus card, while that device is the output destination of the AD8000. Previously recorded tracks can be monitored, along with tracks currently being recorded. If you hold down the AD8000's Monitor button, the headphone display changes from showing the headphone level to displaying one of letters A, B, C or D, to indicate the AMBus being monitored. You can select one of these by holding down the

AD8000 rear panel, with the empty slots for the optional plug-in cards (right).

controlled from within Pro Tools, making it ideal for use with these systems. The only downside I could find is that it can be a little tricky to set up when using it for the first time. Still, after a few days of working with it (and a couple of calls to Apogee) I got mine working smoothly enough.

Project studios running Pro Tools 24 with an 02R mixer should strongly consider buying one AD8000 and the new Digidesign ADAT interface. This combination allows transfer of 24 tracks of audio digitally to and from the 02R, at a cost for the interfaces of about £5000. The alternative would be three 888I24 or, possibly, three 888 interfaces, but these would work out more expensive. The

"The unit sounded extremely good to my ears when recording into Pro Tools 24 or on to my ADAT at 24-bit resolution in bit-splitting mode."

Monitor button and pressing the headphone volume switch up or down. In this mode, all outputs (AES, S/PDIF, headphone monitor, optional 2- or 8-channel D-A converters and all installed AMBus cards, other than the one selected by the Destination function) receive audio from the output of the selected card. To make this clearer, note that the eight output channels always go directly to the four AES/EBU XLRs on the back of the unit. Also, the pair that's being monitored, as determined by the headphone select switch, is routed to the S/PDIF connector, and if an AMBus card contains one or more output channels, the digital outputs that are sent to the AES/EBU connections will be sent out on the AMBus card outputs as well.

These comprehensive routing features let you transfer audio from any connected device to any other, which effectively means that the AD8000 can be used as a powerful multi-channel digital format converter or AES/EBU distribution unit, providing conversion from any interface to all others simultaneously.

SUMMARY

The AD8000's user interface, designed by famous producer and engineer Bob Clearmountain, has certainly been very well thought out, and once I got used to it I began to appreciate just what an excellent job he's done. You can control the stunning array of functions using the front-panel buttons, but many of the settings can also be

advantage with the AD8000 is that you get even higher-quality converters than those used in the 888I24, along with an incredible range of options for interfacing with digital multitracks and other equipment. And the AD8000 is not just for Pro Tools users. At the recent AES convention in Amsterdam, it was announced that the SADiE system would be supporting the AD8000. Steinberg also recommend the AD8000 for use with their forthcoming *Nuendo* software running on Silicon Graphics computers, via an ADAT interface.

In action the unit sounded extremely good to my ears when recording into Pro Tools 24 or on to my ADAT at 24-bit resolution in bit-splitting mode. It also sounded significantly better than my 888 interface, even when I recorded at 16-bit resolution. Highly recommended! **SOS**

PRICES

Apogee AD8000	£4523.75.
8DA 8-channel 24-bit DAC	£1175.
2DA stereo 24-bit DAC	£411.25.
Video card	£323.12.
AES card	£634.50.

ADAT, TDIF, Pro Tools. and Fiber DX cards £411.25 each.

All prices include VAT.

The Final Word In Digital Dynamics Processing

THE NEW dbx DDP

Every Dynamic processor you'll ever need; Compression, Gate, Limiter, De-esser and More with the quality of ultra-wide dynamic range 24bit A to D convertors but with TSE™ (Tape Saturation Emulation) for the warm overload characteristics of analogue tape. Add to this total peak and VU meter coverage of ALL parts of your signal, a choice of Stereo or Dual Mono I/O. AND 50 factory and 50 user presets to remember your settings. And it's by dbx...
...Need we say more?

50 factory presets, 50 user programmable presets

Dual mono or true stereo linked operation

OverEasy® with VariKnee™ or hard knee thresholds

All the classic dbx trademark sounds in gating, compression, limiting, de-essing, and sidechain EQ.

Compression controls: threshold, ratio, gain, OverEasy®, auto, attack, hold, and release

Variable Transient Capture Mode™

Wide-ranging gain control allows for +4/-10 operation

8-step analog input and output metering

Digital input and output meters measure internal digital processing levels

Hybrid display technology. Combines the best of graphics, character and icon based displays

Hi-res graph shows composite output vs. input plot of audio signal

Gate controls: threshold, ratio, attack, hold, release

Ultra-smooth continuous Auto mode.

Limiting from -60dB to 0dB, with gain, attack and release.

Ultra wide dynamic range 24 - Bit A/D and D/A converters with TSE™ Tape Saturation Emulation on board

Digital meters show both peak and average levels

High resolution gain reduction metering

Precision control over every parameter

De-ess from 800Hz to 8kHz, vary the amount.

Sidechain functions for advanced filtering applications. Sidechain monitor included

Midi bypassable via midi program changes.

Utils: sample rate, A/D input and output, Midi functions, Sysex functions.

Gate
Set parameters for threshold, ratio, attack, hold, release, and output gain. See the effect of your settings on the graphical display, as well as on the gain reduction and audio level meters, they all interact in real time with your manipulation of the parameters. Start with a threshold setting of about -60dB to clean off the noise in between the vocal takes.You can save your final gate settings as a "gate preset" building block and recall it into any other setup you do.

Compressor
The effects of the gate settings are visible on the graphic display to help you determine where to set your compressor threshold. Move through all the regular parameters (displayed in real time), like threshold, ratio, attack, release, and output gain. For vocals use a threshold of about -25dB, a ratio of about 3:1 or 4:1, and a slow attack and fast release for the most natural sounding effect. Your compressor settings can also be saved as a building block to be called up into any other preset.
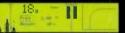
Limiter
Changes you make to the limiter settings are also seen on the graphical display. You can adjust the level and also the speed at which the limiter lets go of the signal as it goes below the threshold. This is truly smooth limiting, with patented dbx PeakPlus™ algorithms, so rest assured that wherever you set your threshold level, your tape will not distort. And like the other parts of the processor, your limiter settings can be named and saved for later recall.

De-esser
De-essing works the same way; see the effects of your settings displayed on the graph. Parameters here are the common ones: threshold (800Hz to 8kHz), and amount (%). Other processing includes EQ - both in-path and sidechain - for special-effect types of processing. When you are editing any of the building blocks, its icon is visible on the display, and the parameters are shown on the graph, so it's always easy to know where you are.

and More
You can also work in stereo, or set up a completely different and independent processing chain for the other channel. Optional digital output with the TYPE IV™ Conversion System with TSE™ (Tape Saturation Emulation) provides up to 24-bit output in either AES/EBU or S/PDIF formats with the trademark digital processing of TYPE IV™. The DDP also has full MIDI/Automation capability, with separate midi in and thru jacks.

Dynamic Range...

266XL • Compressor /Gate • 2 channels of classic dbx compression • Auto Dynamic™ attack and release • Program-adaptive expander gates

1066 • Dual Channel Compressor /Limiter/Gate • V2™ VCA • Hard knee or Overeasy® compression • PeakStopPlus™ limiting

160S • Stereo Compressor/Limiter • From industry standard 165 • High precision input stages • World's largest dynamic range (127dB) V8 VCA

MC6 • Minicomp • Full stereo with True RMS Power Summing™ • Auto mode for vocal or instrument • Full 8 stage metering • Great for studio or live • Switchable hardknee or Overeasy™ compression

286A • Studio quality Microphone Processor • 5 dynamics processors in one • Compressor • De-esser • Exp/Gate • Balanced/Line level inputs

1086 • Mic Preamp/Processor • 48V Phantom power • Optional digital I/o • Variable frequency low-cut filter • VU meter

786 • Solid State Mic. Preamp • Purest source reproduction electronically possible • 130dB dynamic range • 104dB S/N ratio

586 • Dual Vacuum Tube Preamp • 2 channels • Hand selected & matched 12AU7 valves directly in the signal path • Drive control • Peak Plus™

World Radio History

KORG *D8*
Digital Recording Studio

Turn your

Presenting the KORG D8 Digital Recording Studio. Everything you need to make and mixdown your own CD-quality multi-track recordings, easily and without using signal-robbing data compression. It's packed with on-board effects, a built-in 1.4Gb hard drive and a whole lot more. It's an all-in-one personal Digital Recording Studio that will turn your creative dreams into reality.

And all this power comes at a remarkably affordable price too ▾ so there really is no reason to compromise.

Plug Straight In

The D8's input connections accept anything from keyboards and samplers to guitars and microphones, and its S/P DIF input lets you record from DAT or CD digitally, keeping the original sound quality. The built-in sample rate converter ensures everything inside the D8 is at the CD-compatible sample rate of 44.1kHz. At mixdown, you can use the main and aux inputs to add an extra four channels.

Stacks of Built~In Effects

In addition to independent EQ settings for each channel, the D8 has a powerful built-in digital multi-effects processor featuring 50 different high-quality effects. 65 of the effects programs are specially created for electric and acoustic guitar, as well as basses, drums, vocals, sax, etc. Another 65 user locations let you store your edited versions of these. The D8's 12 channel 4 bus mixer allows the effects to be used as inserts during recording or as master effects during mixing.

Record to Built~In Drive

All this capability would be of little use if there was nowhere to record to, so the D8 comes with its own built-in 1.4Gb hard drive as standard, giving over 4 hours of recording time. This equates to more than 2 hours of stereo or around 35 minutes of 8 track. If you want to expand on this, the built-in SCSI interface lets you attach up to 7 external fixed or removable drives, not just for back-up but for direct recording as well.

creative dreams
into reality

DAVE LOCKWOOD tests this hi-tech guitar system which seeks to replicate the sound of many of the most coveted vintage and modern amplifiers on the market, with full digital multi-effects thrown in, and all in a convenient, portable, combo format.

For a guitar amp to make it into the pages of *Sound On Sound*, it has to have something special to offer in the area of technical innovation, and also has to incorporate the speaker emulation circuitry necessary for DI recording. The Millennium One Fifty from the Johnson Amplification division of Digitech scores on both counts, utilising a hybrid configuration of an analogue (both valve and solid state) preamp with digital filtering (for amplifier 'modelling'), and a stereo digital effects stage, followed by a solid-state stereo power amplifier. The whole package is housed in something resembling the traditional guitar combo format, with two 12-inch speakers, and is designed to integrate all the power and flexibility of a sophisticated rack system into the convenience of a portable 'single-box solution'. Despite the inevitable complexity of the resulting system, every effort seems to have been made to keep the Millennium as conventional as possible for the guitarist used to more traditional rigs. Yes, of course there is an LCD screen and lots of pages to scroll through, should you so desire, but the Millennium does not require you to. You can, if you

wish, use the thing more or less like a normal amp, driving everything from the front-panel knobs. Of course, this would waste an awful lot of the power and functionality of the Millennium, but it could be a good place to start for the truly technophobic guitarist who wants to dip a toe in the water before being obliged to dive all the way in.

The two speakers are fed by separate, solid-state power amps (2 x 75W) allowing true stereo effects without the need for an external cabinet, whilst the preamp features two 12AX7 valves, as well as a solid-state gain stage which can be used in parallel with the valve stage. The analogue signal is converted at 20-bit (128 x oversampled) resolution for the effects stage, while a 24-bit internal data path (48-bit with the DSP co-processor) ensures that the signal is not significantly degraded during processing. Up to four digital effects can be accessed simultaneously and all effects and amplifier parameters are available for real-time MIDI control. Parameters may also be assigned to dynamic (signal level-dependent) control, allowing some highly expressive, yet intuitive possibilities, such as chorusing on your clean sound which automatically disappears when you turn up your guitar volume to take a solo, or stereo echoes which do precisely the reverse, keeping your clean sound rhythmically

Virtually Vintage

JOHNSON MILLENNIUM GUITAR AMPLIFIER

uncluttered, but sliding themselves in behind your lead lines. These are very musically powerful effects which would go entirely unappreciated if they weren't easy to program. Fortunately, like the Millennium's external MIDI control assignments, they are every bit as easy to drive as the rest of the amp. A further attraction for some potential purchasers might be found in the fact that, as a software-driven system, the Millennium is software updatable (both patches and the operating system) via the MIDI port — indeed, some of the points I raised with the company at the start of the review were fixed before the end of it with a new software version. Both impressive and comforting if you have invested faith and money in a system of this type.

IN CONTROL

The front panel sports two banks of five rotary controls arrayed on either side of the Master Volume and Contour knobs in the centre of the front panel. The latter provides a useful overall tonal tweak to the power amp — a bit of extra brightness to the right of centre, more neutral to the left. A pair of input jacks, one Bright (and, as always, with a tad more gain), one Normal, will hold no mysteries, but the Input Level pot is the first departure from regular practice. This is simply for optimal matching to the following stages, the intention being to avoid illuminating the accompanying Clip LED except on the heaviest of power chords. About two-thirds up seems to match a stock Strat, whilst a Les Paul is happier at just under halfway and anything with active pickups probably ought to go in the less sensitive input. Clipping this stage only increases your chances of generating some digital crunching further down the chain and is neither useful nor nice sounding. Having matched the input sensitivity to your guitar's output, the next five controls — Gain, Treble, Mid, Bass, Level — seem comfortingly familiar, and operate just as any guitarist from the latter part of this century would expect. To facilitate the Johnson's ability to reset itself totally for each preset, the pots are actually 'rotary shaft encoders' (ie. pots without end-stops) with accompanying two-digit numeric displays to represent the control's 'position' or value. The displays are clean, and actually easier to read on a darkened stage than a conventional knob pointer.

The next challenge is a bank of six switches. Five of these are used to determine what amp the Millennium is pretending to be. The emulations are sensibly divided into family groups, with the choices accessed by successive presses of the switch. Thus, under 'American Combo' we find a mouth-watering choice of '65 Black Face Twin, Twin Reverb Brite, Matchless DC-30 (1), Matchless DC-30 (2), Boogie Mark II Combo, and a Boogie Rectifier Combo. From that, you'll probably have a pretty good idea what you might find under 'British Stack', 'American Stack' and 'British Combo', but perhaps rather less as to what to expect under 'Johnson'. The Johnson type differs from the others in not 'modelling' specific amplifiers — many of the Johnson-based presets in fact incorporate two different amp models running side by side to produce a more complex, composite tone, as well as offering generic sounds like 'Overdrive' and 'Fuzz'. Some of the Millennium's most impressive

sounds are to be found in this group, and I particularly enjoyed being able to 'morph', or crossfade between two significantly different amp sounds using an expression pedal.

There are 36 'models' in all, with two distinct characters for each, portrayed as Channel A and B variants of each amp. These A and B versions are allegedly derived from modelling the real characteristics of those amps which do actually have two distinct channels, with 'clean' and 'dirty' versions of the same amp sufficing for those that don't.

HIGHLY EFFECTIVE

The Millennium organises its sounds into presets — 100 Factory (few of which you will ever want to use straight out of the box, I would contend, but very useful as starting points), and a bank of 100 User memories for storing your own creations. A Millennium preset consists of the selected amplifier and all its associated effects settings and controller assignments. The right-hand side of the front panel is devoted primarily to the effects section. The range of effects on offer here is about as comprehensive as a very superior, dedicated multi-effects unit. The key to the Millennium's success, however, may well lie in how efficiently the non-technical user can tap in to that power, and here I think the designers have done a particularly good job. As with the amplifier section, five rotary controls, designated Mix (overall wet versus dry), Speed (of modulation effect — chorus, tremolo etc), Depth (of modulation effect), Delay (level), Reverb (level), give you instant access to the primary parameters of any effects that are active within the preset. This allows you to always achieve an instant remedial tweak in a live situation, and indeed to drive the Millennium very effectively long before you have learned how it really works. This is important. Guitarists like to think they understand amps, and there is nothing like being able to achieve the basics on first meeting for giving you the confidence to want to explore further. Deeper editing of the effects can be achieved with equal simplicity via a bank of switches dedicated to giving you instant access to the first edit page of each of the main effect groups. It works beautifully and avoids any sense of having to dig around in the depths of the operating system looking for the bit you want to edit.

As soon as you have altered a parameter within a preset, the Store light illuminates in an unmissable red, prompting you to save the change — something which is easily achieved by pressing the Store button three times. Why three? Once to give you a chance to edit the name, once to choose the memory location, and one to confirm. Of course, if you are tweaking a preset without changing the name or memory location, which is what happens most of the time, you quickly end up hitting Store rapidly three times in succession without even looking, as soon as you have done anything you want to keep.

The large, backlit LCD normally shows you the preset number and name, along with a neat block diagram of the elements used and their relative positions in the signal chain. Every preset starts with the option of an analogue wah pedal. To use it in the conventional manner obviously requires the presence of a MIDI control pedal, but you can,

"...the Johnson Millennium scores very heavily in the key areas of sound, flexibility and ease of use."

Johnson Millenium

The preamp controls (above) and effects and preset control area (right).

of course, choose to just use it as a static band-pass filter to replicate the classic 'half-cocked wah' sound beloved of certain rock players. Whilst nothing but the real thing seems able to truly offer the complex resonance of a Cry Baby into a Marshall, the Millennium's 'virtual wah' is a great deal better than most and I could certainly live with it.

The Wah feeds the amp simulator stage, which is followed by a Noise Gate (Silencer I and II, offering a choice of position in the chain), with adjustable Threshold, Attenuation, Attack and Release. An external send/return loop (at -10dBu, a bit hot for most guitar pedals, which are quite likely to be the one thing you'd want to put in here) completes the analogue part of the signal chain.

DELICIOUS DIGITAL

The Millennium's digital effects can be deployed in any one of 15 different configurations, allowing for just about every possible combination of series, parallel, mono, dual-mono and stereo. The full list of 'digital modules' incorporates several different versions of most types of effect, allowing the available processing power to be divided up in the most appropriate way for each preset. Thus, if you were setting up an effect consisting mainly of chorusing, but you wanted to add just a little bit of low-level delay as well, you could still use the Millennium's delicious, but very processor-hungry 'Octal Chorus", which uses three quarters of the total processing available, at the same time as a stripped-down, basic delay program which only needs a quarter. The operating system will only show you a listing of those effects that will fit in the 'slot' you are editing, which forces you to choose the configuration before you know precisely what effects you want to fill it with. This is OK when you are working with the manual, as I obviously was at the start of the review period, but is not so good without it — I sometimes might have chosen a different configuration if I had known which other effects modules it would have made available. The alternative, an operating system which automatically spreads the available power across whatever effects you have selected, is probably easier to work with, but would, of course, prevent you from making exactly the kind

of priority choice in the earlier example. Overall, I think the designers have succeeded in rendering the complex task of organising such a flexible and sophisticated system relatively painless. Included within the digital effects is quite a decent stereo compressor, which can be used to keep levels constant for recording or give the amp a bit more 'feel' at low volume without changing the sound — a compressor at the front-end for deliberate 'squeeze' effects is perhaps the only significant absentee from the regular palette of sounds. One thing you do have to bear in mind when using significant amounts of reverb and delay is that there is no effects 'spillover' when you change patches, making transitions sound rather abrupt unless carefully timed.

SIMPLY AWESOME

In common with most 'processor amps', your initial response to the Millennium may well be determined by the volume level at which you get to try it out. At 'domestic' levels, some of the presets are very impressive indeed. At stage levels, however, those same presets can seem excessively ambient and frequently far too distorted as well. This is an inherent problem with this type of amp — the compression, distortion and EQ that you select in order to sound like a loud amp being driven close to its limit does not translate particularly well to actually being loud! For example, Factory Preset 1, 'Rectifier Solo', sounds simply awesome when you sit in front of it 'noodling about' (guitarists will know what I mean), but tends to simply disappear when used alongside a real bass player and drummer. It's not that the Millennium can't go loud enough, for it can be screamingly loud (particularly on the Boogie/Soldano-type presets) but as a consequence of the way that heavy distortion inherently deprives the signal of dynamics and the way the artificial ambience combines with the real room ambience to push your sound into an imaginary distance. Winding down the delays and reverbs to less than a quarter of the programmed value in many instances, knocking back the amp models' front-end gain (not the input matching sensitivity) and trimming a bit off both the top and bottom to compensate for the ear's greater sensitivity to these frequencies at high levels produced something far more pleasing, much more convincing and indeed, far more usable in a band context.

I did find, however, that there was a significant disparity in level between some of the presets, depending on the amplifier model used. If the 'amp model' output level is already at maximum, the only way to balance the presets against each other is by trimming the level of the louder ones. Where this is done to a preset that has already had its front end turned down, the overall output volume can start to become a problem. The Millennium may have two 75-Watt power amps on ▶

THE EFFECTS LIST

Dual Chorus
Quad Chorus
Octal Chorus
Dual Flanger
Dual Phaser
Rotary Speaker
Stereo Tremolo
Auto Pan
Dual Detune
Quad Detune
Octal Detune
Smooth Pitch Shift
Dual Pitch Shift
Quad Pitch Shift
Octal Pitch Shift
Stereo Pitch Shift
Stereo Dual Pitch Shift
Harmony
Delay
Dual Delay
Quad Delay
Stereo Delay
Stereo Dual Delay
Stereo Quad Delay
Long Delay
Analogue Delay
Stereo Analogue Delay
Chorus Delay
Flange Delay
Pre Delay
Reverb
Dual Reverb
Stereo Reverb
Gated Reverb
Stereo Gated Reverb
Room Echo
Spring Tank
Auto Wah
Compressor
8-band Graphic EQ
15-band Graphic EQ
31-band Graphic EQ
Stereo 8-band Graphic EQ
Stereo 3-band Parametric EQ
6-band Parametric EQ
Stereo 6-band Parametric EQ

World Radio History

Johnson Millenium

board, but they will only be driving about half of that into each of the internal 8 Ohm Celestion speakers. With those presets that I had already tweaked for level matching and to get rid of excessive distortion, I found the overall volume control creeping up towards maximum without giving me the same kick that I have come to expect of my normal valve combo (Mark III Boogie). Sixty-odd Watts into two twelve-inch speakers ought to be loud enough to keep up with a bass combo and un-miked kit, so the real answer to maximising the performance of this amp in a live context relies on the manufacturer addressing the limited volume of the quieter presets.

Amplifier systems which generate all their distortion at the preamp stage are inherently more noisy than those which achieve a significant amount of it within the power amp. The Johnson falls into the former group, and you probably will want to use the noise gate if you are playing at reasonable volume, using distortion and a few effects at once (the effects themselves are not noisy at all, but they do amplify the preamp noise). I found both gate types a little unsubtle and would have preferred a dynamic filter process which left some noise, but crucially some signal still present, whilst removing the most intrusive hiss component. For recording, I preferred to leave the gate out altogether, deferring the choice of anti-noise strategy until mixdown and making sure that decaying notes were never prematurely chopped off.

DIRECT RECORDING

For the majority of *Sound On Sound* readers, the DI recording outputs will be among the most important features of the Millennium. Digitech's speaker emulation will be well known to many guitarists from products like the 2112 processor, and performs as well as most, with the added advantage that all the additional processing that is always necessary to achieve something reasonably convincing is right there within the system. The function, which operates only on the balanced XLR DI/Line Outputs, may be switched in and out, either globally or on an individual preset basis — it perhaps is worth taking it out for anything completely clean and sparkling, or if you happen to like distorted sounds that will drill holes in your teeth, otherwise, personally, I'd leave it in all the time.

The nominally +4dBu, DI output level is actually determined by the Master Volume control setting.

For recording, where you probably only want to hear the sound via the control room monitors, the manual recommends that you disconnect the internal speakers. This is perfectly OK, of course, for a solid-state power amp, but it's not exactly elegant, and I am sure that some less technical users brought up on valve amps will not be all that comfortable doing it! Frankly, on a system of this sophistication, I can't see why the designer chose not to have a separate recording output level control. Its absence is even more of a limitation in a live situation where you are effectively using the internal speakers just as a local monitor while your PA feed is taken from the DI, for it is impossible to vary your monitoring level without affecting the FOH feed. I appreciate that that's exactly what happens with a miked-up speaker, but why emulate a limitation when it would have been quite simple to improve on it?

For recording, just as for live use, many of the Millennium presets are far too ambient to be used without editing. The obvious 'quick fix' approach of tweaking the overall wet/dry mix, however, actually varies the balance of the analogue (front-end and amp simulation) and digital (effects) stages, which means that the chorus and pitch-shifter will disappear along with the reverb. The 'analogue/digital balance' control configuration confers the advantage of allowing you to set up a basic 'dry' amp sound and then add processes to it, but chorusing and pitch-shifting become an integral part of any sound on which they are used and once the required balance is achieved, you will not want to vary it. The facility to be able to quickly balance the dry sound, including any modulation or pitch-shifting effects, against the reverb/delays would, though far more complex to implement, be infinitely more useful, both on stage (to compensate for varying acoustics) and in the studio (allowing a quick re-balance of just those effects you would probably prefer to leave until the mixing stage).

The quality of the effects is quite excellent and on a par with anything else that it might make sense to use within a guitar rig. The only one I couldn't come to terms with at all was the rather bizarre spring simulation. Mechanical spring reverb, for all its limitations, is particularly suited to certain clean guitar sounds and still well worth simulating in the most sophisticated of digital systems. While most simulation designers take the opportunity to de-emphasise the less desirable characteristics in favour of the more useful ones, the Millennium's spring simulation appears to consist *entirely* of the less desirable characteristics! Perhaps users can look forward to a software upgrade on this one.

IMPRESSIVE

Despite my criticisms of some aspects of its operation, I really enjoyed using the Johnson Millennium during the test period. Like all 'emulating' amps, whatever their principle of operation, it is not actually anything like using the real thing, but some of its (intentionally) less subtle sounds are nevertheless very impressive in their

own right and a joy to use in the correct context. Many of the high gain presets 'feel' very right indeed, allowing you the full range of expressive pick technique variations. Pinched harmonics are a doddle, and on all but the flat-out thrash presets, the sound cleans up nicely as you back off the volume control. Inevitably, the classic 'valve amp just going into overdrive' sounds are more of a compromise, but that is in the nature of the system and will be well accepted by the kind of user I imagine the Millennium appealing to — the player who has to be able to totally re-invent their sound from song to song, or perhaps even from verse to chorus, at the same time as needing a portable, self-contained system — perhaps the covers band player, the busy session player who carries their own gear and, of course, the home recording player who also gigs. There are other products in the 'super amp' market, but the Millennium's nearest rival, in terms of the breadth of its capability, is probably the AX2 212 from Line 6 (see Paul White's update in this issue, page 40) and if you are in the market for either one of these, you should make the effort to check out the other as well. This is a highly subjective area where judgements can not be made from a comparative features list.

If you are a guitarist who needs a lot of effects and you require the maximum amount of flexibility

that technology can currently offer in both your recorded and live sound, you are looking at two choices: a custom component system, with preamp, power amp, multi-effects and tuner in a rack, plus a couple of cabinets (for stereo); or one of the current generation of 'super amps' offering all of the above in a portable package. If your preference is the latter, then the Johnson Millennium scores very heavily in the key areas of sound, flexibility and ease of use. **SOS**

info

£ £1499.95 including VAT.
A Arbiter Group, Wilberforce Road, London NW9 6AX, UK.
T +44 (0)181 202 1199.
F +44 (0)181 202 7076.
W www.arbitergroup.com
W www.digitech.com

"The quality of the effects is quite excellent..."

HARD CHOICES & SOFT OPTIONS

PART II: Following on from last month's look at the advantages and pitfalls of analogue and digital tape, and hardware tapeless recorders, PAUL WHITE turns his attention to the ways in which computers can be used in audio recording.

CHOOSING A RECORDING SETUP

Computers have been part of mainstream studio equipment ever since *Cubase* and *Creator* first became available for the Atari ST, but it's only in recent years that they have become powerful enough, and hard disks cheap enough, to make audio multitracking on a PC or Mac both practical and affordable. I'm not going to touch on the Mac-versus-PC argument in this article, but rather consider the merits or otherwise of entrusting audio to *any* type of computer. The ability to integrate and control everything in a single environment is very attractive, as is the facility to cut and paste audio information in the same way as MIDI data, but what about the horror stories — the hardware incompatibilities, the timing problems, the crashes, clicks and buzzes? And even if everything works perfectly, will the computer do everything a tape-based system did? More importantly, will it do it without interrupting the musical process?
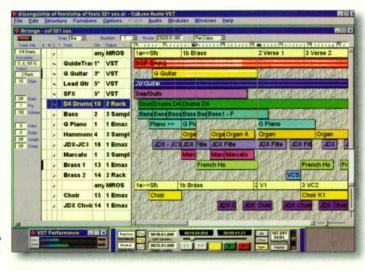
AUDIO WITH COMPUTER'S OWN HARDWARE

Perhaps the biggest growth area in contemporary recording is music on computers. Most of us use computers as MIDI sequencers, but the majority of sequencers now also include audio support that relies entirely on the host computer's processing power. PC users will need to fit an audio card to get sound into and out of the computer, whereas Mac users have the advantage of the 16-bit stereo analogue I/O built as standard features into all the latest Macs.

Computers have grown significantly more powerful in terms of both speed and processing power over the past two years, while RAM prices have fallen to less than 10% of what they used to be. This has allowed software designers to use some of the additional processing power to provide real-time effects and mixing, so that in theory you can have 90% of your studio entirely inside the computer, including perhaps one or more MIDI sound sources. Given enough processing power, you can even throw in software synthesis!

So-called 'MIDI + Audio' software packages like *Cubase VST* and *Logic Audio* provide MIDI sequencing, multitrack recording, digital mixing with EQ, plus bundled and third-party software effects

Cubase Audio VST v3.5.

plug-ins. Recording is limited to no more than two channels of audio at a time on a Power Macintosh, and the same for a PC fitted with a basic stereo-in, stereo-out soundcard. Because the effects, processing and mixing are all internal, having only stereo outputs may not be a problem, but whether or not you can live with recording only two channels at a time depends on your requirements.

You can see why this approach is so attractive — it's cheap, it sits on a single desktop and it offers advanced features such as automated mixing, graphical editing of audio files, and manipulation of audio tracks using the same familiar tools with which you manipulate MIDI tracks. However, those who've used such systems will know that it's not all plain sailing.

To start with, you need the most powerful computer you can get your hands on if you're not to run out of horsepower when patching in virtual effects and processors. There'll always be a limit on what your computer can do, but on slower machines, that limit is reached very quickly. In a hardware-based studio, all the processors are always available for use – but in the virtual world, effects and processors only exist so long as there is sufficient computing power to run them.

Assuming you have a fast machine, what other problems can you expect? Firstly, unless you buy a MIDI control surface with physical faders, you're going to have to do all your mixing on-screen using the mouse. You can automate the mix and keep tweaking it until it sounds right, but it's still a lot more tedious than using a hardware mixer. If this doesn't worry you, ask yourself whether you can live with the physical noise of a computer and a fast hard drive. As computers get faster, the processors run hotter and the cooling fans have to be stronger. The result sounds like an amplified fan heater, which is an unwelcome intrusion both when recording and mixing.

Maybe you're happy with all of these potential drawbacks; but now we come on to reliability. Firstly, you'll have to back up data when the hard drive is full, so you need to buy a removable media drive. What's more, even computers that are set up as well as possible will occasionally throw a fit, so if you're working in a commercial or time-sensitive environment, ask yourself whether you can survive a major system crash, which may or may not involve data being permanently lost. Even if your system works fine one day, you may find that the simple act of loading a new piece of software or updating an old one can throw the system into a fit that takes several guru-hours to fix. Even the best-written software has some bugs lurking in the background, waiting to spoil your day. Indeed, you may be one of the unlucky ones who never gets their system to work properly in the first place!

We receive many calls from users whose audio sequencer tracks drift out of time with each other, whose soundcards refuse to talk to their software, whose hard drives (or graphics cards) cause ticking on the audio outputs, or who experience unacceptable delays when playing back audio tracks. The only advice we can give is to buy your computer from one vendor as a complete system, and don't use it for any purpose other than music. If you don't follow this rule, don't be surprised if nobody will help

you out — each individual component will probably check out fine, and it's only when you put them together that you get problems. Also, if you're one of those people who can't resist installing and running games or, worse still, those dubious bits of software that come on magazine cover disks, you might as well sign up for the computer-guru night classes now; you'll need them!

The quality of recordings made using only integral hardware depends largely on the quality of the analogue-to-digital and digital-to-analogue converters used, but as a rule these are unlikely to be as quiet as good outboard converters simply because they're inside the computer. The best audio cards seem to offer a signal-to-noise ratio of about 90dB, while the worst provide less than 60dB (though the manufacturers' figures don't always relate to the performance of their card in your specific computer). You'll need a good mic amp or voice channel to get a quality signal into the system, and if you use external MIDI modules, you'll still need a hardware mixer to combine the output from the computer with the outputs from your MIDI modules. What's more, because the inside of the computer is essentially a closed system, you'll be extremely limited in patching in any of your hardware effects and processors.

COMPUTERS + MULTI-CHANNEL CARDS & INTERFACES

The type of system described above can be made much more flexible by adding a multi-channel soundcard. The cheaper of these have the A-D and D-A converters on the board itself, while the more upmarket models may put the converters in an external box so as to give better signal-to-noise performance. Most low-cost multi-channel cards, such as Emagic's Audiowerk8, have fewer inputs than outputs and are therefore most suitable for people who record by overdubbing one or two parts at a time. If you spend a little more on something like Event's Layla, you can have eight inputs that allow you to record several tracks in one take.

The main advantage of multiple outputs is that you can feed your tracks to separate channels in an external mixer, and this in turn means that you can use your hardware effects and processors more effectively. Indeed, your computer now operates more like a multitrack tape machine as far as the rest of your system is concerned. ▶

Emagic's Audiowerk8 multi-channel PCI card.

AUDIO WITH COMPUTER'S OWN HARDWARE

benefits
- Lots of features in one integrated package.
- Affordable.
- Takes up relatively little space.
- Integrates MIDI, audio and mixing in one environment.
- New software updates bring new features.
- Wiring is kept to a minimum.

disadvantages
- Powerful computers are needed to run multiple effects, processors and mixing alongside audio recording and MIDI sequencing.
- Some form of data backup system (or fast removable-media drive) is necessary.
- New software upgrades may not run on your existing machine, so computers have to be upgraded periodically.
- Computers are physically noisy, while monitors can interfere with electric guitars and basses.
- Patching in hardware effects and processors is awkward or impossible.
- System crashes and hardware compatibility problems can be extremely difficult to sort out.
- The best software packages have a steep learning curve.
- Unless you buy additional hardware, everything has to be controlled from the keyboard and mouse.
- Computers don't always handle manual punching in and out as well as a tape machine or dedicated hardware recorder.
- Can be difficult to transfer project data from one studio to another.

▶ However, you don't have to use all the outputs to carry separate tracks — a program that provides mixing within the computer may let you configure some of the outputs as track or mix outputs and the others as effects sends, in which case you could use a much simpler external mixer and still be able to patch in your hardware effects units. Whether or not you can do this depends on the choice of card and software.

AUDIO ON COMPUTER WITH EXTRA DSP ASSISTANCE

The next step up from running a system in which the computer's own CPU does all the work is to give it some assistance in the form of plug-in DSP cards, as in Digidesign's Pro Tools system, or external hard disk recording hardware such as the Yamaha CBX D5. Newer systems include the Ensoniq PARIS, and there are other well-established players such as the Soundscape SSHDR1 and SADiE. Prices can vary enormously, and the better systems may cost considerably more than the computer into which you plug them — but for work where you need a lot of tracks and a lot of real-time processing, this is the way to go.

I'll use Digidesign's Pro Tools system as an example, because it has been around for a long time and enjoys good third-party support both from sequencer manufacturers and from the third-party software developers who design plug-in effects and processors. Here, the hard work is done by multiple DSP chips located on PCI cards plugged into the computer, and to ensure optimum audio quality the multi-channel audio interfaces are outside the computer in a separate rack box. The DSP chips handle mixing, EQ and plug-in effects, and there's also circuitry dedicated to looking after the transfer of audio to and from hard disk. This takes a load off the computer's mind and generally makes for a more stable, more powerful system.

Another advantage of these systems is that if at any point you need to run more software plug-ins, some of them (Pro Tools included) will allow you to add further DSP cards. Pro Tools also has its own internal bussing system known as TDM, which allows audio signals to be routed in many different ways, and it's via TDM that third-party plug-ins are accommodated.

SUMMARY

At *SOS*, we are frequently asked about what is the best type of recording system to buy. As you

can see from this series, there is simply no such thing, because every approach has its own set of strengths and weaknesses. Computers are incredibly flexible and offer a lot of functionality for relatively little money, but they can be very temperamental, they make a lot of physical noise, and both the hardware and software evolves at a frightening rate.
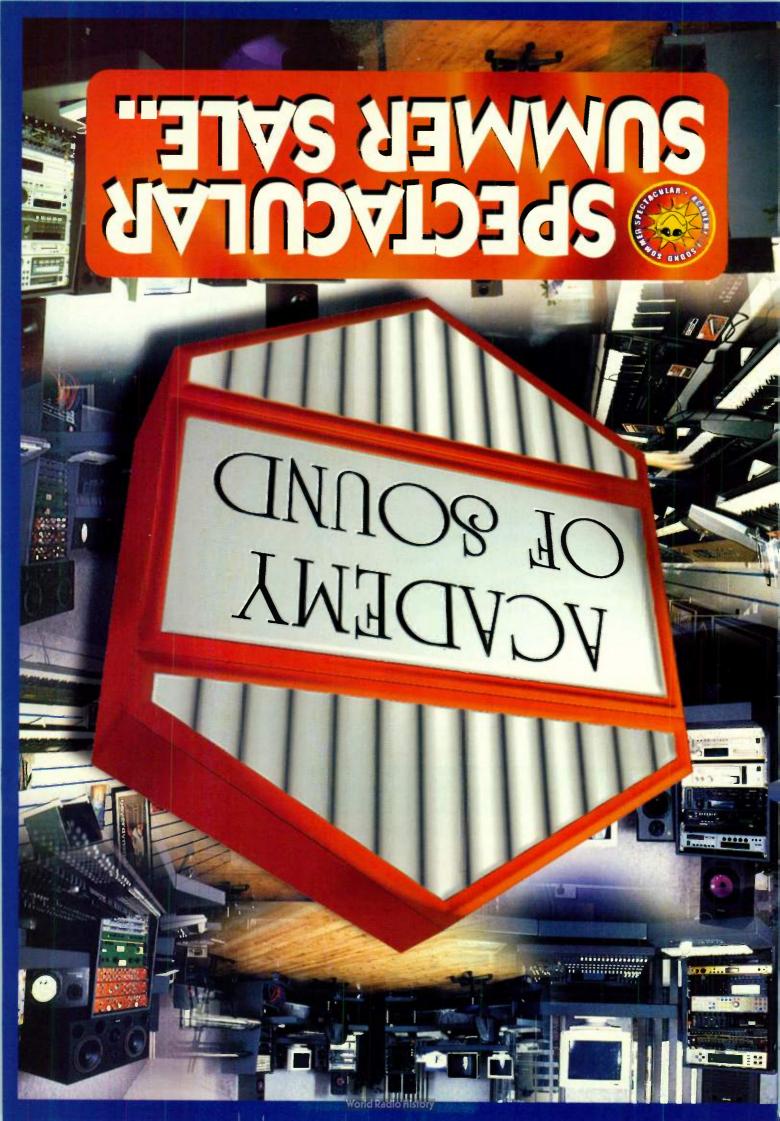
Modular digital multitrack tape (see last month's instalment) offers good compatibility with other studios, and the media cost is cheap, but there's no random access and locking two or more machines together is, to be honest, pretty tedious. All-in-one digital multitrackers (also dealt with last month) have their appeal, but the mixer may not be big enough to handle all your other sound sources, and if the mixer section is digital, you'll probably find that it offers only limited possibilities for plugging in your own analogue processors, or more than one digital effects unit (this is a problem shared by many stand-alone digital mixers).

Analogue tape is reliable and has a musical sound, but it deteriorates gradually with use, so you might find that you've lost a little top end by the time you've finished a project. What's more, on semi-pro machines, the noise reduction can affect the sound quality quite noticeably, and of course there's no random access facility, nor any way to copy a tape or a section of tape without incurring some quality loss. However, sudden and total data loss is unlikely unless the dog eats your master tape.

Ultimately, the right choice is determined by the job you want to do, your budget, how prepared you are to learn something different, and whether you can afford the time to make backups if the system you choose has a fixed hard drive. Everybody has a different way of working and a different set of needs, but hopefully this short series will help point you in the right direction. **SOS**

*The Digidesign
Pro Tools System.*

Long awaited and much anticipated, Novation's new digitally-modelled analogue synth module is the company's most ambitious offering yet. SIMON TRASK has a blast with the Supernova...

Although they established themselves as providers of budget analogue bass monosynths with the BassStation series, Novation have, to date, lacked an instrument to put them in the synthesizer Big League alongside heavyweights Korg, Roland and Yamaha and fellow European upstarts Access and Clavia. While all these manufacturers have brought out polyphonic digital synths utilising physically modelled analogue synthesis, Novation's only foray into sound modelling has been their DrumStation rackmount drum module. Now, however, Novation are launching the Supernova, and for all those people who've been waiting and waiting, yes, it's finally out — first models started shipping around the middle of June. Based on the custom Analogue Sound Modelling digital technology Novation first developed for the DrumStation, this synth module represents a significant leap in terms of pricing, technology and functionality. As well as being Novation's first synth to break through the £1000 price barrier, it's also their first polyphonic, multitimbral digital synthesizer. Yet at heart it embodies the same enthusiasm for analogue synthesis technology that has always characterised Novation. So will the Supernova light up your sky at night, or is it destined to be eclipsed by bigger stars?

OVERVIEW

The Supernova is a 3U 19-inch rackmount module which packs in a traditional waveform-based subtractive analogue synthesis architecture, modelled digitally in software, 8-part multitimbrality, an 8-part arpeggiator, and seven effects *for each part* in multitimbral mode! It comes in two versions, offering 16- and 32-voice polyphony respectively, with the 16-voice model being upgradable to 32 voices. The 32-voice model also has twice the 512 Program and 256 Performance memories of its 16-voice cousin, though the polyphony upgrade for the latter model also includes the additional memories.

Novation's new module has two play/edit modes: Program and Performance. With the module in Program mode you can play a single Program at a time, on the user-selectable Global MIDI channel. Each Program has seven effects, including reverb, chorus and distortion, and can be assigned a preset or user-programmed arpeggio drawn from a common memory pool. To use the Supernova MIDI-multitimbrally, you must select Performance mode; this gives you eight Parts, each of which can play a single Program complete with all its effects. You can assign each Part to either the Global MIDI channel, Omni (ie. all), or one of channels 1-16. As you can also set a key range for each Part, this means you can create anything from a single-channel split/layer texture of up to

Star Attraction

NOVATION SUPERNOVA ASM POLYSYNTH

eight Programs through to an 8-channel MIDI multitimbral configuration with one Program per channel. In Performance mode you can also use up to eight arpeggios at once — one assigned to each of the eight Parts, triggering a single Program with associated Part-specific effects processing. Individual Parts, complete with any effects processing, can be routed to any one of four pairs of audio outs on the module's rear panel; you can use any of these eight outputs as individual outs by panning one or more Parts hard left or right.

SUPER STYLING

Memories of the BassStation's less than sturdy construction and its rather toy-like appearance haven't done Novation any favours in subsequent years. However, the Supernova is set to lay those memories to rest. The new module is solidly professional in both appearance and construction, and at the same time sleekly and distinctively stylish, thanks to its metallic blue casing, deep-blue fluorescent LCD (á la Lexicon) and plethora of red pinpoint LEDs and custom-designed knobs and buttons. Both the response and contours of the knobs and buttons are reassuringly firm, yet smooth and comfortable, helping to give the Supernova a plush feel. Novation have also managed to pull off their usual trick of balancing a generous controller-rich front panel on the one hand, and a clearly organised and accessible layout on the other, with knob and button positioning that's just on the right side of cramped. The user interface has been thoughtfully designed to make parameters as accessible as possible, via a combination of knobs and buttons and LCD menus; LCD-based parameter editing is straightfoward and fast, thanks to a combination of page up/down buttons and a Fast Data knob plus a pair of increment/decrement Nudge buttons for each of the two LCD rows.

SYNTHESIS

A Supernova Program's sound sources consist of three main oscillators and a pink noise source. In addition, each of the main oscillators has its own 'virtual' slave oscillator for creating sync effects, and there are also two ring modulators which take their inputs from oscillators 1 + 3 and 2 + 3 respectively. For each of the three main oscillators you can select a square or sawtooth waveform, and set pulse width for the former; Novation plan to add further waveforms in a future update, making them accessible via the Special button in the Oscillators section. You can adjust the octave (+/-2), semitone (+/-12) and fine pitch (0-127) amounts for each of the three main oscillators, but not for the noise source, which is a shame. Incidentally, OS updates are uploadable into the Supernova via MIDI from a sequencer as MIDI files; an uploaded file will be loaded into temporary RAM, checksummed for load errors, then copied into Flash ROM if OK. Internet users will be able to download updates for free from Novation's web site.

The outputs from the (sync'ed) main oscillators, the noise source and the two ring modulators are routed through a mixer section, where each source's level can be set independently. Each level can also be modulated by any combination of five sources, with a separate modulation amount being specifiable

The Supernova Oscillator section offers Waveform, Tune and Modulation controls.

The filter section.

for each source. The five sources — LFO1, LFO2, envelope 2, envelope 3, and mod wheel — form one side of a 5x5 modulation matrix, with level, pitch, pulse width, sync, and 'soften' (more on this in a moment) forming the destination parameters. Only the level parameter applies to all four sound sources and the two ring modulators.

The standard oscillator sync effect is available for each of the three main oscillators via the sync level parameter in the modulation matrix, with a mod level parameter also available for each of the five mod sources. However, the Supernova also goes beyond traditional oscillator syncing capabilities with three additional parameters, not part of the modulation matrix and so not dynamically controllable: sync skew amount (-64/+63), formant width (0-127), and sync key follow amount (0-127). Together these three parameters — again, settable individually for each of the main oscillators — further increase the timbral modification possibilities available on the Supernova prior to the filter stage. In essence, the soften parameter rounds off the 'sharp edges' of a waveform, which has the effect of reducing the waveform's harmonic content and creating a 'softer' sound; for instance, a square wave can be reduced all the way down to a sine wave. Soften can be applied independently not only to each of the three main oscillators but also to the noise source. In addition, for each one of the three main oscillators you can set the pitch wheel range and direction (+/- 12) and the degree of mod wheel and aftertouch control of LFO 1 modulation amount (- 64/+63 in each case).

The output of the mixer section is, of course passed on to a filter section, which consists of a single resonant filter, capable of being driven into self-oscillation. You can choose between low-pass, band-pass and high-pass filter types and 12, 18 or 24db/octave roll-off. Currently the filter doesn't have a notch filter type, but Novation plan to add this in a future OS update. Other filter section parameters are resonance and overdrive, the latter being a saturation effect which can be used to create a fuller, warmer sound, and there's also a 2x5 modulation matrix, with filter cutoff point and

The Supernova has three ADSR envelope generators and two LFOs.

"The new module is solidly professional in both appearance and construction, and at the same time sleekly and distinctively stylish."

▶ resonance amount as the destinations, and LFOs 1 and 2, envelopes 2 and 3, and the mod wheel as sources (as in the oscillator section). In addition, you can specify separate aftertouch modulation amounts for frequency and resonance, and the degree to which aftertouch and mod wheel individually will affect LFO2 modulation of frequency and resonance. You can also set a filter tracking amount and a Q norm amount; the latter determines to what extent, if any, higher levels of resonance will produce a louder output signal from the filter.

The Supernova has three ADSR envelopes — one assigned to amplitude, the other two assignable. The attack, decay and release stages each have maximum durations of 20 seconds. You can specify the degree of control that attack velocity has over each envelope's modulation depth (controlling the volume of the Program in the case of envelope 1), the amount of delay for envelopes 2 and 3, and the amounts of mod wheel and aftertouch control over envelopes (again, controlling Program volume). Each envelope can be set to single or multi triggering (single prevents legato notes from retriggering the envelopes). There are also several parameters, as follows, common to all three envelopes:

- Portamento type (porta or glissando — continuous or semitonal gliding).
- Portamento mode (linear or exponential glide — exponential is used for TB303-type portamento).

- Poly mode (determines whether or not a repeated note will use the same voice).
- Oscillator mode (percussive or ensemble — determines whether or not the oscillators are reset on note on).
- Glide type (normal, autoglide or eight pre-glide types).

Regarding the latter parameter, autoglide is used for TB303-type glides, with no portamento on non-legato notes, while the pre-glides provide initial up or down pitch glides, with a choice of two-, five-, seven- or 12-semitone ranges, and duration determined by a portamento time parameter.

The LFO section consists of two LFOs, each with speed and delay settings and a choice of square, saw, triangle and sample-and-hold waveforms. The Supernova's LFOs can operate at audio frequency rates, a feature which lets you create sounds not possible with the standard LFO speed range. For convenience, an LFO Range parameter lets you set whether the Speed knob will operate over Slow (moderate), Normal (standard synth) or Fast (full) ranges. You can alternatively sync one or other or both LFOs to Internal or MIDI clock, depending on the Global sync setting, and select a 'note duration' for each LFO, ranging from a 32nd-note triplet up to 12 bars; for internal sync, the rate is set by the Supernova's arpeggio speed, and for MIDI sync by the external sequencer's tempo. In this way you can use the LFOs to create rhythmic effects sync'ed to a MIDI sequence.

Other parameters available per LFO are offset amount, envelope 3 LFO speed mod amount, aftertouch and mod wheel LFO speed mod amounts, single or multi delay trigger, and keysync or freewheel LFO triggering (ie. whether or not the LFOs are in phase and reset with each new note played). Many of the modulation amount parameters mentioned above have a +/- value range, so you can determine not only the amount but also the direction of modulation — so, for example, more aftertouch decreases the amount of modulation.

EFFECTS

Each Program can draw on up to seven effects modules: distortion, stereo reverb, stereo chorus/flanger/phaser, stereo delay, comb filter, EQ, and stereo panner. At present they're organised in a single configuration, with distortion, EQ, comb filter and panner organised in series, while reverb, chorus/flanger/phaser and delay effects operate as individual send/return effects (with send amounts settable individually) after the comb filter and before/after the panner. However, Novation are planning to add multiple configurations in a future OS upgrade.

For the distortion effect you can simply set drive amount (0-127) and mod wheel amount (-64/+63), while for the EQ you can set treble and bass boost amounts (-64/+63), and for the comb filter, the filter frequency (0-127) and comb boost (-64/+63). The send/return effects offer more programmability. In the case of the delay, for example, you can set ▶

World Radio History

TWIDDLING OVER MIDI

Novation first championed MIDI sound edits with the BassStation, albeit in a limited way. With the Supernova they've really gone to town, making extensive use of both MIDI controllers and Non-Registered Parameter Numbers to transmit data from all of the module's 28 knobs and 98 buttons, plus all but one of its LCD menu-based parameters (the Restore parameter in Global mode). NRPNs were devised as a way of extending the MIDI controller map of 128 controllers. MIDI controller numbers 98 and 99 are the NRPN LSB and MSB respectively, allowing for a total of 65,536 controllers! MIDI controller number 6 (the Data Entry controller) is then used to edit the value of the parameter indicated by the NRPN LSB/MSB number. Novation use only the LSB, giving 128 controllers, but they use the data values of some of these NRPN controllers to provide value ranges for multiple Supernova parameters. The entire list of parameters and values takes up seven A4 pages in the manual. Of course, all this parameter data can be recorded into a sequencer via MIDI, for subsequent playback to the module, or it can be used to edit another Supernova via MIDI, while anyone who wants to develop a synth editor module or a sequencer mixer map will have an absolute field day with the possibilities on offer (Novation themselves will be offering some mixer maps for free download from their Web site).

Arpeggiator

The arpeggiator has a total of 384 patterns.

delay time, feedback amount, stereo width, left/right delay ratio, delay signal high-frequency damp, mod wheel control amount, and sync. Like the LFO rate, the delay time can alternatively be synced to internal or MIDI clock, with a note length ranging from 32nd-note triplet to 12 bars. With the panner you can, of course, set a fixed pan position (L63-R63), but alternatively you can specify a pan type (autopan, tremolo, L-R, R-L), pan speed, pan depth, and dry level (controlling the amount of source signal that gets through). Overall, the effects are capable enough, and definitely a worthwhile addition to the instrument — all the more so for being Part-specific in Performance mode. While they're not always the richest or smoothest effects you might hope for (the reverb, for instance, has a ringy, warbling quality when it's laid on thick), they integrate with and augment the analogue quality of the instrument to good effect.

PERFORMANCES

As I said earlier, a Performance consists of up to eight Programs, which are assigned to eight Parts. MIDI channel reception and (for the arpeggios) transmission for all eight Parts defaults to the Global channel, as defined in Global mode, but you can assign each Part to a different MIDI channel if you want, for MIDI multitimbral use. There are six settings buttons in Performance mode:

- Tune: allows you to set semitone (+/-24) and fine-tune (+/-63 cents) transpositions for individual Parts.
- Range: assigns key ranges to Parts (within C0-C8).
- MIDI: sets the send/receive channel for the selected Part.
- Velocity: lets you set a response curve (normal, inverse, crossfade high, crossfade low) or a 'from or to' numeric value (from various preset values — a freely settable value range would have been better), so there's scope for velocity crossfades and switches.
- Output: allows you to route each Part *with its effects*, as described earlier.
- Polyphony: sets each Part to Poly, Mono or Off.

When you select a Part, using the relevant Part Edit button, all the synthesis and effects settings for the Part's Program are made available via the relevant front-panel knobs and sliders. You can fully edit the Program in Performance mode within the context of the other Programs assigned to the Parts. Then, when you come to save the Performance, after you've chosen a destination memory and entered a Performance name you're asked whether or not you want to update the Programs. The default value is No, but alternatively you can select Yes or Each. If you select Yes, any edited Programs are automatically written to the relevant Program memories. However, if you select Each you can go through each Program one by one and decide whether or not you want to update it in memory. At this point you can select a different Program memory to Write into; in

A front-panel section labelled 'effects' provides an adjustable control knob and a menu button for each of the seven effect-types.

addition, the Compare button lets you listen to the Program stored in the destination memory — an extremely useful feature.
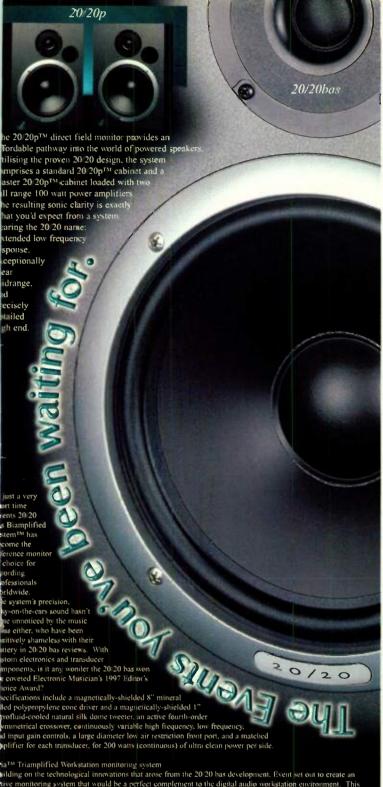
As with the Program Banks, all Performance Banks are writable, and Novation have left Bank B empty in the 16-voice model, and Banks B-D empty in the 32-voice model, so there's plenty of free memories for storing your own patches and edits.

Novation have trounced the competition by allowing Programs to keep all their effects and associated settings when assigned to Parts in Performance mode. For each Performance Part you can use the assigned Program's effects settings or create your own specifically for the Performance. Unfortunately, however, you can't copy effects settings from a Program into the Performance memory, so 'tweaking' becomes a laborious process. This shortcoming is something Novation plan to address in a future OS update.

You can mute and unmute individual Parts live using the eight Part Edit buttons (a feature which really comes into its own with multiple arpeggios). Just press the relevant button 1-8 to select the Part, then each subsequent press switches the muted/unmuted status. What isn't possible is muting or unmuting of more than one Part at the same time, which is disappointing for live mixing. However, the Supernova is apparently able to read multiple Part buttons at once, and Novation are thinking of introducing different button-press options to make multiple Part muting possible.

ARPEGGIATOR

The Supernova provides 128 preset monophonic and 128 preset polyphonic arpeggios, and allows you to create a further 128 arpeggios of your own (64 mono and 64 poly). In Program mode, each Program can be assigned an arpeggio from this common pool; you can also program the tempo (64-191bpm), whether or not the arpeggiator is automatically enabled when the Program is selected, and whether or not it's latched (continues to play when you take your fingers off the keyboard). In Performance mode, these settings are per Part within a Performance (except for tempo, which is the same for all Parts), either adopted from the Part's assigned Program ▶

SUPERSPEC

SYNTHESIS METHOD:	Analogue Sound Modelling
POLYPHONY:	16 voices (expandable to 32) or 32 voices
OSCS PER VOICE:	3 oscillators with virtual sync, plus noise generator
MULTITIMBRALITY:	8 parts
FILTER:	12, 18, 24db/octave, resonant; low-/high-/band-pass options; overdrive control
LFOs:	2 assignable LFOs capable of operating at audio rates
PROGRAM EFFECTS:	Distortion, Comb Filter, EQ, Reverb, Chorus/Flanger/Phaser, Delay, Panning
TOTAL EFFECTS:	56 available simultaneously in Performance mode (8 Parts x 7 effects each)
PROGRAMS:	16-voice model: 512 (expandable to 1024); 32-voice model: 1024
PERFORMANCES:	16-voice model: 256 (expandable to 512); 32-voice model: 512
ARPEGGIATOR:	384 patterns (128 preset poly, 128 preset mono, 128 user); each Part has its own arpeggiator in Performance mode
AUDIO OUTS:	8 assignable polyphonic outs, 24-bit 128x oversampling delta-sigma DAC each
MIDI CONNECTIONS:	In, Out, Thru
MIDI CONTROL:	All knobs, switches and parameters transmit MIDI controllers and NRPNs; arpeggiator can transmit via MIDI; arpeggiator rate, LFO speed, and delay time can sync to MIDI clock
POWER SUPPLY:	Built-in, multi-voltage, automatically switches for different voltages and currents
DIMENSIONS:	3U 19-inch rackmount; 133mm x 222m (including knobs) x 483mm (HxDxW)
WEIGHT	3.5kg

▶ or else programmed within the Performance itself. Other Program- and Part-specific arpeggiator parameters include number of playback octaves (1-4), gate time (25, 50, 75 or 99%), arpeggio direction (up, down, or as played), key reset on/off, mute on/off, latch type (repeating or one-shot), note length (32nd-note triplet to 12 bars), and velocity response to trigger notes. For monophonic arpeggios you can also set how the Supernova

"The Supernova marks Novation's arrival in the big league."

handles situations where the number of trigger notes is greater than the number of different notes in the arpeggio sequence. Another feature of the Supernova lets you transpose any part of a played arpeggio in real time, using notes sent on a user-programmable Arpeggio Transpose channel; this feature can be turned on or off for individual Parts in a Performance. You can also choose, for each Program and Part, whether arpeggio notes play internally, via MIDI, or both; in Program mode, arpeggio notes are transmitted on the Global MIDI channel, in Performance mode on the channel of the relevant Part. Individual arpeggios can be dropped in and out in Performance mode by muting and unmuting the relevant Parts — though, as mentioned earlier, disappointingly only one at a time.

If the arpeggiator is to be used in conjunction with a MIDI sequencer, the Supernova must be set to External sync so that it will lock to the sequencer. With External sync enabled, the

Supernova reads MIDI Start, Stop and Continue commands, along with MIDI clock data. The arpeggiator tempo is clocked to the sequencer's tempo (and therefore to any sequenced tempo changes). However, you can use the Arp Sync function (available per Part in Performance mode) to change the base note length of an arpeggio, so that it's possible to have different arpeggios playing at different rates and looping at different times. In practice, the Supernova's arpeggiator starts, stops and syncs to a sequencer accurately, both in an extended linear play situation and when you have a sequence loop running repeatedly.

Last, but by no means least, user arpeggio programming is done in Global mode. Here it's possible to select one of the 128 available user arpeggio memories, choose a sequence length of 1-64 steps, and assign a note number, velocity amount and gate time to each step. Providing that the arpeggio you're creating is assigned to the Program in use, you can get instant feedback on your step-time editing efforts simply by playing trigger chords on your keyboard or looping an arpeggio trigger note or chord in your sequencer as you work.

CONCLUSION

The Supernova marks Novation's arrival in the big league. It's a versatile and feature-packed analogue-modelled synth that convinces sonically, functionally and visually. Eight-part multitimbrality, an 8-part arpeggiator, seven effects per part, and a maximum 32-voice polyphony help to put the Supernova on the map as a serious challenger in the 'new model analogue' market. But what really mark it out as a classy synth are its seductively rich, warm, smooth and natural analogue-modelled sound, and the character, versatility and sheer playability of its Programs and Performances. Combine all this with an accessible, controller-laden front panel for easy editing, and a sophisticated MIDI implementation for effective integration with today's sequencer-based recording setups, and you have a traditionally-styled synth module for the late '90s and beyond, that will appeal to a wide range of hi-tech musicians and quite possibly become a workhorse instrument in project and professional studios alike. A cosmic supernova is a short-lived phenomenon. This Supernova, on the other hand, looks destined be a bright star in the synthesizer firmament for some while to come. 　SOS

World Radio History

Renowned producer and remixer Pascal Gabriel is back with a stylish new recording project, Peach. With a hit in the States behind them and the inclusion of a Peach track in the film Sliding Doors, widespread recognition can't be far away. BILL BRUCE meets Pascal and production partner Paul Stratham at their studio to talk technology, and finds them going back to their fruits...

S hadowed by the pristine monolith of Exchange Square, where city types sip champagne around an astroturf croquet lawn, the crumbling brickwork of former factory and industrial spaces is being turned to more creative uses. It is here, in the increasingly fashionable EC1 area of London, that the Strongroom complex of programming rooms and studios is to be found; and this, in turn, is where Peach blossom. Peach, of course, is the collective name of the stylish musical entity comprised of production team Pascal Gabriel and Paul Stratham, and vocalist Lisa Lamb. Though the name may not be familiar, you may well have heard their song 'On My Own', which is featured in the recent hit movie and Gwyneth Paltrow vehicle *Sliding Doors*, and which dented the US Top 40 at the end of last year. Together, the group have just released their debut album, the unambiguously titled *Audiopeach*.

The idea behind Peach was to produce big unashamed pop moments, although the members all arrived at this concept via very different routes. Lisa is a published poet who came from the jazz and R&B

Life's a Peach

PEACH • PASCAL GABRIEL & PAUL STRATHAM

world, having worked with Troublefunk and Washington DC outfit Band of Gypsies. Paul, meanwhile, emerged from the more gloomy reaches of the rock scene. After the dissolution of his own band, the early '80s new wave hopefuls B-Movie, he spent some time working with the King of Goth, Pete Murphy (ex-Bauhaus). While working in Istanbul on their last album together, Paul met producer Pascal Gabriel.

Pascal has come the furthest in distance, although the sound of Peach is not very far removed from that of the work he has produced over the past 10 years. At 16, he left his native Belgium for London partly to avoid conscription into the army, but also due to the fact that, Plastic Bertrand aside, the Walloon pop scene was not exactly vibrant. Working with Mark Moore of S'Express fame, and Tim Simenon of Bomb The Bass, he co-wrote and produced some of the recordings that were instrumental in bringing dance music overground in the late '80s. Pascal then went on to produce or remix several artists on the independent label Mute (now also home to Peach), from Erasure and Depeche Mode to the Inspiral Carpets.

JUICY FRUIT

Meeting Pascal and Paul in the café bar of the Strongroom, we are treated to the Air album as we drink coffee and chew gum. In a way the Gallic electro sound of the background music is not a million miles away from Peach's vibe. Both groups share the same Euro-style kooky/cool ambience, and the Peach sound is also typified by synth sounds instantly identifiable as belonging to another era. However, instead of Air's sparse analogue '70s-style meanderings, *Audiopeach* is more reminiscent of the epic '80s synth pop of The Associates or to a greater extent, ABC, with an upbeat retro feel similar to St Etienne.

Audiopeach was recorded in Pascal's home, and much of the equipment from that setup now resides at the group's Strongroom studio. Located at the top of a flight of stairs, the studio, while fairly compact, still has the feeling of light and space, due in no small part to the colour scheme; walls and ceiling are painted blue with little fluffy clouds. Pascal explains the idea as he shows me the black vocal booth (not pictured); "The vocal booth is the night sky, with the little pearls in it, and the other room is the sky during the day. Instead of having another boring old studio I wanted to make it look a bit nicer." A stack of vinyl is propped against the wall, comprising everything from House 12-inches to Nancy Sinatra and Lee Hazlewood, from The Stooges and MC5 to those kitsch *This Is Phase Two Stereo!*-type albums so beloved of charity shops. Half of one wall is taken up by the large screen of an Apple Mac, an abundance of synth rack modules and three Roland System 100 modular analogue systems, with the impressive tangle of patch leads that entails. If this closely resembles the setup on which *Audiopeach* was recorded, it's clear that it wasn't put together in just any old bedroom studio.

Pascal: "We used Emagic's *Logic Audio* for sequencing from day one, together with Digidesign's Pro Tools; we have eight tracks of audio on that, with two I/O cards and we run tons of MIDI stuff and samplers, an Akai S3000 and an Emax. Everything on the album ran through the Mackie desk — we did all of the monitor mixes and even a couple of the final mixes through that and straight to DAT; all the B-sides, remixes and club mixes. All the 'proper' mixes and all the mixes for singles were by Dave Bascombe at Abbey Road, Eden and the Townhouse. We would record the tracks, do the mixes the best that we could at my place, then dump them onto multiple ADATs and let Dave finish them off. But all the recording itself was done at my place."

PERSPECTIVE & HOW TO LOSE IT

Many musicians aspire to the kind of relaxed recording setup Pascal and Paul have access to. But as Paul explains, the working method is not without its problems: "Although having your own recording space means you don't have to complete a session within a set time frame, there are still different kinds of pressure. You get pressure from your record company and publisher, and you put pressure on *yourself* to get something out. It's two years since we made this record, and ▶

Left: Peach (l-r): Pascal Gabriel, Paul Stratham, Lisa Lamb.

Left: The Strongroom setup. Left to right, top to bottom: ARP Odyssey & Sequential Circuits Pro One synths, Mackie 32•8 mixer, Korg Mono/Poly analogue synth (on the floor), Dynaudio monitors. In the rack on the right are the Empirical Labs compressor, the dbx 120X DS Boom Box, an Alesis D4 drum module, the Mutator Mutronics filter bank, an Oberheim Matrix 6R, and the Roland SPV355 pitch-voltage synth.

Below: Keyboards and digital recording: Roland Juno 106S & JD800. The floor pod contains the Opcode Studio 4 MIDI interface, Digidesign Pro Tools hardware, I/O units, and SMPTE Slave Driver, plus a Tech 21 Sansamp speaker simulator. A Vox Bass can also be seen on the right.

▶ now it's out it does sound to my ears a little dated. So the pressure is on yourself to come up with something contemporary and quick."

Paul goes on to offer one explanation for the delayed release of the album "I think what happened was that 'On My Own' started to happen in different territories at different times [see the 'Away Win Syndrome' box]. So everybody started hanging back on releasing the album, as if saying 'Let's see what it does there...'.

I wondered why the group had opted to get in other people, in other studios, to do the final album mixes; surely this had contributed to the delayed release, and also ran counter to their working methods on the rest of the project? However, it seems that the reasons were more artistic than technical. Pascal explains: "I've mixed a couple of records that I've produced at my place, and really we could have finished the Peach album here. But we had grown too close to it by the time we came to the rough mixes, so we really wanted someone else to finish it off. We had a lot of instruments and parts going at the same time; we filled pass after pass of the ADAT's eight tracks. We didn't mix down to one ADAT tape, we'd just dump it all on to several with the timecode on track eight, and run it off in blocks of seven tracks. Some of the songs needed 48 tracks. You just forget when you run so much MIDI stuff together how many tracks you are using." Paul revels in Peach's excesses: "There is a lot going on on the album. We tended to double a lot of things up; we'd take a string part and copy it into so many banks... I think that's what gave the album the feeling it has."

After putting so much detail into the tracks, Pascal came to the conclusion that they couldn't see the wood for the trees. "That's why we got people like Dave Bascombe in to mix; by the time we recorded the tracks I couldn't see what should go where balance-wise. I knew it was good, but I knew it had to be sorted out by someone who could look at it objectively. As a producer, I can do it for other people, but I can't do it for my own songs. Because if you've written it, produced it and recorded it, you know it so well that all of it sounds important to you. It's easier to give it to someone else — and listening to what they do to it is also exciting."

ON MY OWN — THE 'AWAY WIN' SYNDROME

In common with Radiohead and more recently Cornershop, Peach have found success in the United States first, going top 40 and receiving substantial airplay with 'On My Own'. I asked Paul if he thought that this kind of 'away success first home later' story was a emergent trend; "I think it's entirely a matter of perspective. It's the kind of view you take if you're one of the bands that aren't happening in England! If you look at the UK charts there seems to be a lot of bands popping in and out. It seems to be a much younger audience now dictating who is going onto the chart; every week a new 'put-together' band just crash in and drop out.

"'On My Own' is a weird little song that has really hopped around the place. It's been covered by a Japanese artist, Shazna, and become the theme music to a big Japanese soap opera! And now it's in Sliding Doors..." Pascal enthuses; "Shazna is expecting to sell one and half million copies, which is great. That one song is being covered by all these people! Someone has even translated it into Mandarin for a cover in China." Paul sarcastically muses "We'll have to get someone in England to cover it if we want a hit."

"And if you don't like it, you can tell them, because you're paying them!" interjects Paul waspishly.

CRISP FLAKE OR ANGEL CAKE?

The big sound of *Audiopeach* is achieved with the help of a fair load of vintage equipment. Pascal runs through some of *Audiopeach*'s seasoned veterans: "We used our old synths like the Roland MKS80 Super Jupiter and Juno 106, a Sequential Prophet VS and Pro One, the SE1 [*Studio Electronics'* 'Moog-in-a-Rack']. A lot of our string sounds come from a Yamaha SY77. Then there was the Oberheim Xpander; but by the time I got a patch going on that, I'd forgotten the reason for creating it in the first place!"
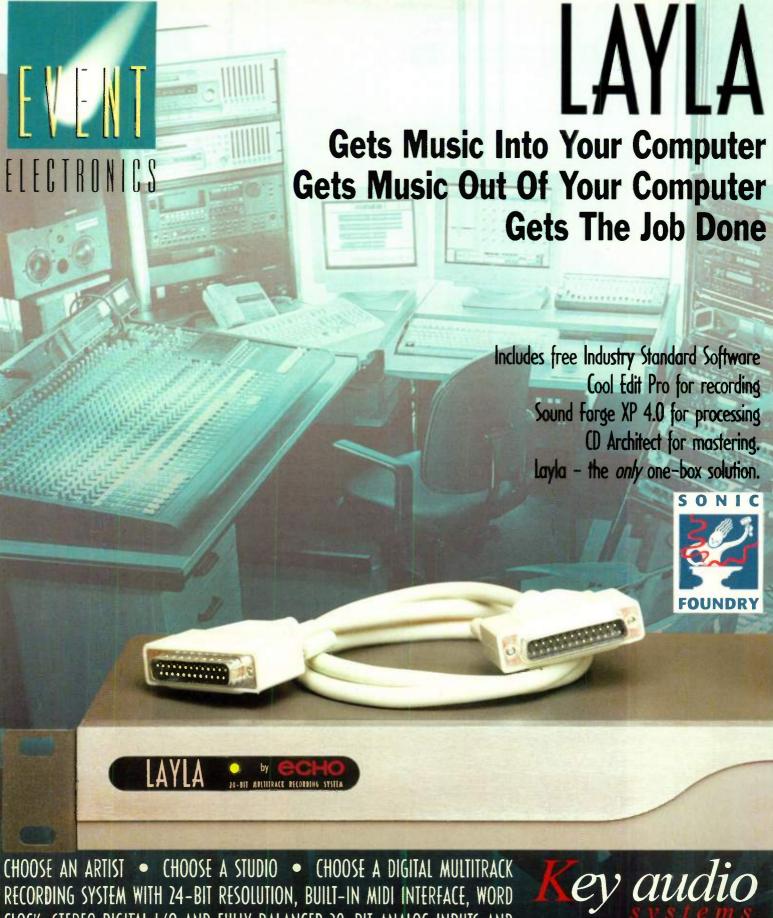
Most *SOS* readers would no doubt agree that Peach's studio is a great setup. Nevertheless, at the moment, Pascal is contemplating a synth car boot sale: "I'm into downsizing; it's all just weighing me down. Sometimes it's a distraction; you think 'We need a good bass sound' — and we've got 15 machines that do those!" Paul chips in, "If you've got one machine, you will work with that and you will come up with more idiosyncratic sounds, rather than going 'Oh, that's a really good preset...'"

Part of Pascal's planned scaling-down of his setup may involve moving more of his studio on to his computer desktop, an idea which intrigues him greatly: "Ideally, what I would like to get is a G3 Power Book with eight tracks of audio and a sampler ▶

Racks in a rack, including (left rack): a Roland MPG80 Super Jupiter Programmer & MKS80 synth, Waldorf MIDI bay, Roland A110 MIDI display and M480 48-input line mixer; (right rack) Studio Electronics SE1 'Minimoog in a rack', Sequential Prophet VS, Clavia Nord Rack, Yamaha TX802, Emu Vintage Keys, Novation DrumStation & BassStation rack, Akai S3000 sampler. The Novation BassStation keyboard is also visible on the right.

built in. You have cards which adapt PCI cards on to a PowerBook so you can have your audio, your sampler, synths or whatever on the one machine. You can have a virtual desk on your Mac, with maybe just one real rack containing all your I/O hardware, but with all your EQs, processing and levels controlled by the Mac. It's a beautiful way of doing it, I feel." But traditionalist Paul introduces a note of dissent. "I tend to disagree slightly, after been stuck in a room using *Cubase VST*. After using the onscreen mixer for a bit, you start to wish you could get your hands on something real." Unfazed, Pascal counters: "Sure, but you can have a MIDI desk with knobs and faders that you can assign to anything if you want to work that way." Paul laughs: "So, you'd strip it all down and then build it all up again!"

Something the duo do agree on is that by trimming back on the amount of equipment they use they will be able to bring out more of the intrinsic qualities of the sounds they do use, rather than stacking up textures to achieve an epic sound. Pascal: "You can't just stockpile sounds on top of each other endlessly. It's like a cake with a thousand layers in it; all the layers blend into one and you can't taste any of them. I'd much prefer a crisp flake now." Paul continues the bakery metaphor: "Or an Angel cake: just three colours and a nice creamy bit. I think you will have to pay more attention to the actual synth you are using, the sound you are creating, so it'll stand out on its own, rather than say something will just about do because you won't hear it anyway, because there'll be a huge string part on top of it.

"I actually think it takes a lot of balls to strip your songs down to the bone. It's like putting yourself on the line. You can fill up multitracks and it sounds great and unusual but when you strip the song down... The old Velvet Underground stuff, for example, sounds so powerful, and yet it didn't take thousands of overdubs. And Joy Division created so much tension and atmosphere. I heard 'Love Will Tear Us Apart' on the radio the other day and it just sounded amazing. You could hear absolutely every instrument; one guitar, one bass, one guy thrashing away at the drum kit and that huge string pad played over the top of it..." Pascal chimes in: "We'd like to do that, in a more electronic style. Yazoo's stuff was really powerful in the same way, because you had a great vocal and really simple music. Not that I want to sound like Yazoo! I think there must be a way to do something of that simplicity that is powerful... that is now." He adds with humility, "But we haven't quite come across it yet."

EFFECTIVE TREATMENT

Now that they are attempting a sparser approach to songwriting, Pascal and Paul are finding the quality of individual sounds becoming more important. Apart from the attention to detail that is employed in the programming and recording stages, Pascal stresses that high-quality outboard gear (of which they've always been fans) is now more important than ever. "Reverbs are reverbs; I'm not really that bothered about them so much. Effects that I find *really* interesting give the sound a different characteristic without it being noticeable. Like a fantastic compressor, for instance, or a really good EQ, preamp or mic. The changes are subtle but you really *notice*. Especially with vocals; there can be no reverb or delay on them, but it can still sound great and all be down to the compression or just a great mic. That's the kind of sound I'm interested in, more than delays, reverbs or other effects, because that can create distance and depth of field." Pascal elaborates on the kind of gear he's talking about: "Usually when I mix, I hire a couple of Fairchild compressors, really old expensive valve compressors which cost about six grand each. Then further on the stereo chain I put in a hired GML (George Massenburg) EQ. That's one of those things where a mix just sounds better when you put it through, even with the knobs at zero." Paul agrees and defines the essential quality that ▶

SELECTED ITEMS FROM THE STRONGROOM SETUP

KEYBOARDS/SYNTHS
- ARP Odyssey
- Clavia Nord Rack
- Emu Vintage Keys
- Korg Mono/Poly
- Novation BassStation x2 (keyboard & rack)
- Oberheim Expander
- Oberheim Matrix 6R
- Oberheim Matrix 1000
- Roland JD800
- Roland Juno 106
- Roland MKS80 Super Jupiter (with MPG80 programmer)
- Roland SPV355 P/V (Pitch-to-Voltage) synth ("That basically gives you a synth noise or CV output to whatever pitch you put in it; it'll follow a drum part or a vocal.")
- Roland System 100 modular (3 systems)
- Sequential Circuits Pro One
- Sequential Prophet VS
- Studio Electronics SE1
- Yamaha TX802

RECORDING
- Alesis Quadraverb & Quadraverb GT
- Ashly Parametric EQ (x2)
- Drawmer DS201 Dual Gate
- Drawmer DL221 Compressor/Limiter
- Dynaudio monitors
- dbx 120X DS Boom Box bass enhancer
- Empirical Labs Distressor compressor
- Mackie 32•8-channel desk
- Mutronics Mutator compressor
- Panasonic SV3700 DAT machine
- Roland SRV2000 digital reverb
- Tech 21 SansAmp guitar amp simulator
- Watkins Copycat valve tape echo unit
- Yamaha NS10M monitors

SAMPLING
- Akai S3000

DRUM MODULES
- Alesis D4
- Novation Drumstation

COMPUTER & SOFTWARE
- Apple Power Mac
- Emagic *Logic Audio*
- Digidesign Pro Tools + SMPTE Slave driver

MISCELLANEOUS
- Akai CD player
- Opcode Studio 4 MIDI interface
- Roland Midi Guitar (used just as a guitar)
- Vox Bass
- Waldorf MIDI bay

erious about

music

Full Time Study

▶ **Higher Diploma in Contemporary Music**

▶ **Diploma in Contemporary Music**

- Funding & Finance Available
- Fully Accredited
- Limited Places Available for September 1998

Professional Opinion:

"This is the first programme I've seen that really addresses the challenges faced by today's top producers" (Steve Harris, producer - Kula Shaker, The Bluetones, Longpigs)

Course consultants include:

- Kevin Nixon (Manager, Kula Shaker)
- Wez (Mushroom Records-Garbage, Ash)

Call us today!

for your Free brochure & enrollment details

01483 456788

f a x : 0 1 4 8 3 4 5 6 0 7 0
91 Haydon Place, Guildford, Surrey GU1 4LR

▶ **Music Production & Technology**
▶ **Guitar**
▶ **Bass**
▶ **Drums**
▶ **Vocals**

The Academy Of Contemporary Music

Overview:

The Academy of Contemporary Music is more than simply tuition, it is about creating opportunity for musicians by combining world class training with unparalleled insight into how the Music Industry functions.

Facilities:

Situated in Guildford city centre, the schools operate from two specially converted buildings that offer a 16 track digital / 32 track hard disk recording studio, a dedicated music production suite with the latest PC hardware and software, four fully equipped teaching studios, a concert hall, resource centre and modern reception.

"The best equipped music school I have ever seen" (Dave Widdecombe, Fender Guitars)

Instructor Profile:

All ACM instructors are world class performers, composers, arrangers and technologists. Our Music Production & Technology instructors and consultants have experience working for the BBC, Top Studios, Commercial Radio, Theatres, The Music Press and have worked on numerous Top 10 albums and Top 40 singles. In addition to this instructors from our Guitar, Bass, Drum and Vocal schools have toured & recorded with top professionals including: George Michael, Gabrielle, Lisa Stansfield, Jools Holland, Bryan Adams, Chris Rea, Guns & Roses, Bon Jovi and Aerosmith giving our music technology students unique insight into high end studio and live production.

Full Time Study:

The school offers full time study at two levels, the Diploma in Comtemporary Music and the Higher Diploma in Contemporary Music. Both programmes have been written by some of the countries top producers, writers and session players to formalise a study path for students wanting a realisitic and intense training which will prepare them for a professional career in the Music Industry. Students studying at ACM are able to form a strategic approach to their career drawing on advice from our course consultants including Kevin Nixon (Manager, Kula Shaker), Wes (Mushroom Records - Garbage, Ash) & Steve Harris (Producer, The Bluetones, Longpigs). These programmes are fully accredited and are run in collaboration with Guildford College of Further & Higher Education.

Free Consultation:

All students applying to study at the school are entitled to a 'Free Consultation Session' to evaluate their musicianship skills and career objectives ensuring that they select the most appropriate course.

An aging Roland SRV2000 reverb shares rack space with an Alesis Quadraverb and Quadraverb GT, Drawmer DS201 Dual Gate & DL221 Compressor/Limiter, 2 Ashly SC63 Parametric EQs, Panasonic DAT, Akai CD player.

HOT STUFF — TAPE ECHO

One of the more unusual bits of kit in the Peach studio is an old Watkins Copycat tape echo with a large silver knob bolted on to it. Fascal is obviously a fan. "It's great, isn't it; it's a really old valve one. It originally only had an on/off switch so you couldn't make it go faster or slower. So we had to try three kinds of dimmer switches and the first one was really good, it had really fast response. It could go from really fast to really slow within a quarter-turn of the dial — but it burnt out in about an hour! So we had to finc one that would take about six months to burn, calculating the ratio of speed of burning out to reasonable response. It's basically a mains dimmer that controls the speed of the motor. It works really well, but it does get a bit hot."

▶ top-drawer gear gives to your recording. "It's true. Just switching it on, you smile and you think 'that sounds like a record!'"

Since the dawn of digital recording, people have sought a way of bringing the 'warm' quality of analogue recordings into the digital realm. Pascal has employed various techniques ranging from the conventional to the more off-kilter. "I've got this other compressor called the [*Empirical Labs*] Distressor; a mono single-rack unit which is *really* great for making things sound 'analogue'. It's fairly severe; when you put stuff through it, it sounds really aggressive, almost like tape saturation on an overloaded tape machine. Sometimes, I'll use samples that I've put down on cassette or even on my Dictaphone. We've put drums looped through the Dictaphone because it's got a compressor on the mic; it's a horrible cheesy little mic with a 4K bandwidth. But it's exciting, and it sounds great. I don't have *any* tape machines apart from my Dictaphone!"

NO TAPE PLEASE, WE'RE PEACH

Ah yes. Pascal has *never* been a fan of analogue recording. The idea of laying down audio in a format which is (unless you are willing to fiddle about with tape splicing) carved in stone doesn't appeal to him. He goes on to make a rather eyebrow-raising statement, considering he has been recording for over a decade: "I have never, ever, had tape in my studio. I just used loads of samplers when I first started 10 or 11 years ago. I had *Pro 16* on the Commodore 64 and I ran everything live and just sampled the vocals! I always liked the idea of being able to manipulate the lead vocals with the track; after recording, you can often think 'I wish he had pushed that chorus a bit more'. With minus delay, it's done in seconds. Before I had hard disk recording, I would do it with samples and just trigger the vocals a bit earlier... I've always been into having full control of everything that's going on. I love the way you can record, dump everything into your computer, send the band away, and be left to fiddle with the bits until you are happy with it."

This dedication to digital recording means that Pascal embraced hard disk recording early on, while others were hanging back to see how well the new medium coped with its teething troubles. "I've been satisfied with hard disk recording on the Mac during the last four years, but before that it was chancy. I started with [Steinberg's] *Cubase Audio*, and I had a lot of trouble with that. That's why I went over to *Logic*, actually, because before I got the Mac, I had been a *Notator* user on the Atari, which was rock-solid, a really cool sequencer. I went over to *Cubase Audio* and it really wasn't ready at the time. There were all sorts of problems, like audio delay on playback. You'd record your audio and on playback it would be 20 milliseconds earlier and out of time with your MIDI stuff. I would call them up and they would say 'Oh yeah, we know about that.' It didn't mention it in the manual, though..."

Unsurprisingly, Pascal uses digital editing quite extensively as a creative production tool as well as for correcting timing and pitch, coming up with some of the group's more off-the-wall sounds this way. "We do a lot of analogue filtering and then further chop that about digitally — I've got an ARP Odyssey with an external Audio In so you can trigger the filter from the external input. We also do live recordings of Paul on synths messing around with the filter and resonance controls, making all sorts of strange sounds for a few minutes. You record all this stuff, and then you can take just a bar, or two beats, or take a section and reverse it. Or put a part from the verse into the chorus that's in a different key, and then need to transpose it up."

THE PARTNERSHIP: RIFFMEISTER & MIXMEISTER

Despite this unashamed and extensive use of technology in the production process, Pascal and Paul insist on keeping the songwriting stage of the process simple. They cite Bacharach & David, Serge Gainsbourg and even Cole Porter as songwriting influences, and so unsurprisingly, when it comes to composing, they leave the technology behind. Paul: "We sit down with a guitar or a good piano sound and we actually write 'songs'! The sequencing comes later, because we have found if you try it the other way round, you can go for days and days, building up this monstrous track... and then you realise there is no song! It's nicer to nail the song first. Then you can relax and have more fun with the production side." Explaining the dynamic within their partnership, Pascal adds, "We don't do the backing track without the song, and the melodies go down early. Paul comes up with the chord sequence and I will come up with a melody for it. He's the Riffmeister as well." Paul adds, "Yeah, but obviously Pascal has a lot more experience than I do in the balancing of sounds... if I'm the Riffmeister, he's the Mixmeister. Basically, whatever one person can't do, the other person can. We've not had the need for anyone else to come in." Pascal and Paul: the archetypal modern, self-sufficient music-making unit.
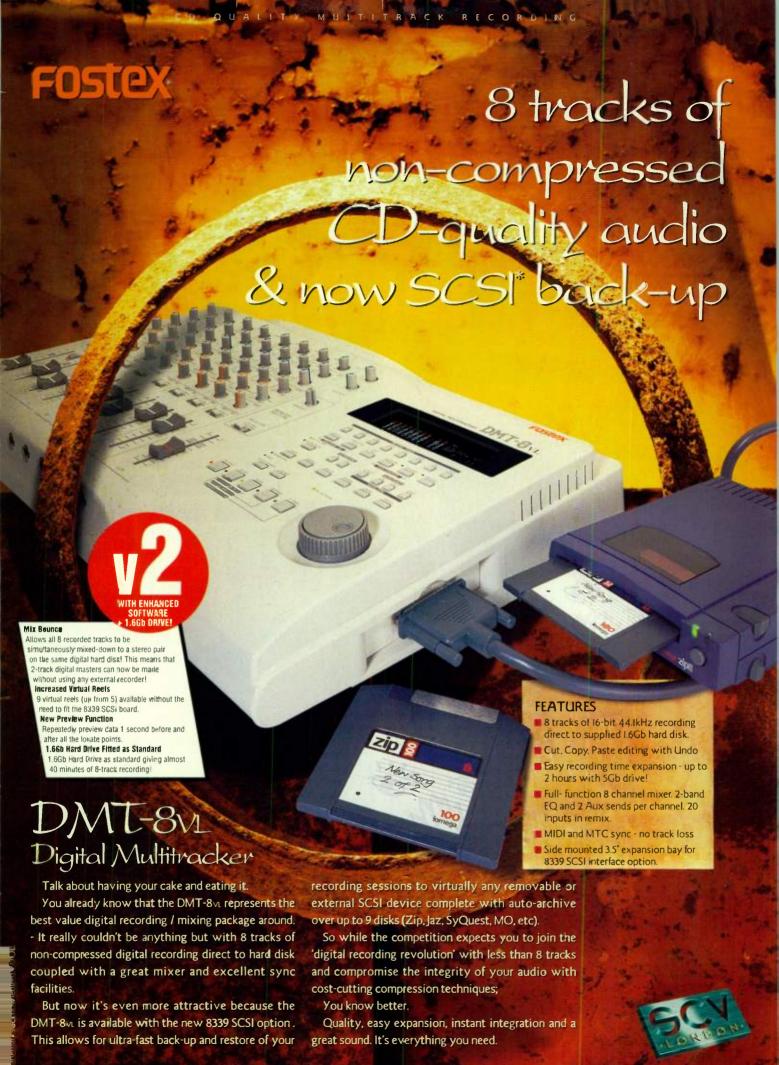
THE FUTURE'S BRIGHT... THE FUTURE'S PEACH
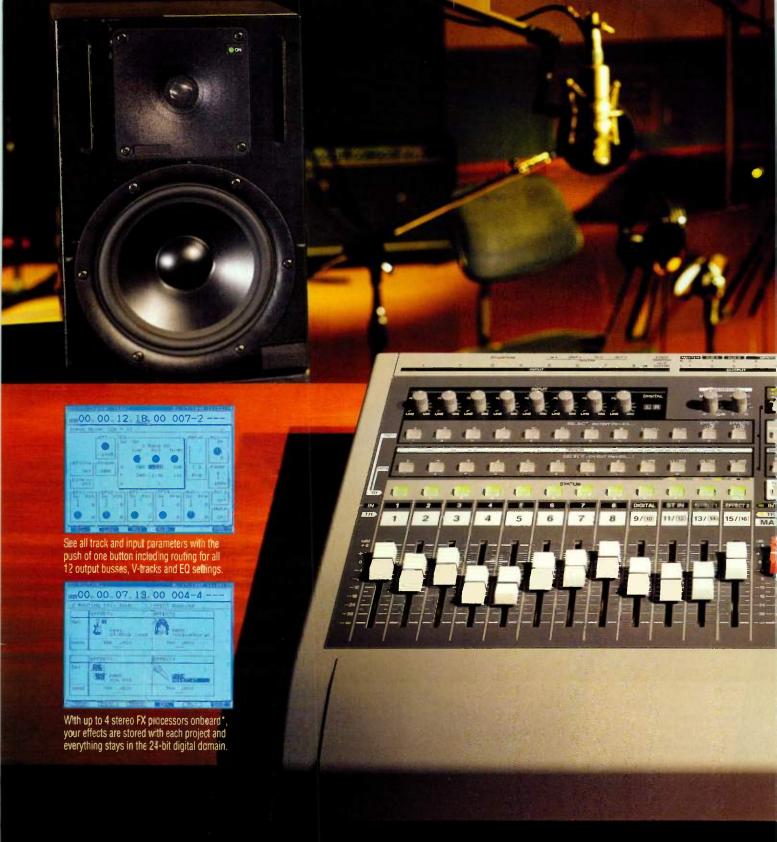
The immediate future includes pondering how to approach live work. "We did a few gigs last year on Erasure's tour, although that was just using an ADAT, playing live over gaps we had left on it. But if we strip down and write music that is not as 'big' then it'll make the live side much more accessible. On the other hand, if you spend your time thinking 'how can we do this live?', it restricts how you write." The next Peach album is also ongoing and more work as a writing and production team working with other artists is also a priority.

Looking around the studio packed with various goodies, I pose them a final question: your studio is on fire and you can only save one thing; what would it be? Paul lets out a tortured "Aaargh!" and then settles for his Telecaster, while Pascal plumps for the ARP Odyssey. Paul: "Then we'd have to do an album with just a Telecaster and an Odyssey... we'll have to do a track on the next album with just that now." Perhaps that's taking the minimalist approach a bit too far? [SOS]

World Radio History

See all track and input parameters with the push of one button including routing for all 12 output busses, V-tracks and EQ settings.

With up to 4 stereo FX processors onboard*, your effects are stored with each project and everything stays in the 24-bit digital domain.

Studio One: The VS-1680

Finally, your dream digital studio is within reach. With its 16-track playback (8 simultaneous record) and 256 virtual tracks, up to 4 stereo (or 8 mono) built-in effects processors *, 26-channel automated digital mixer, MT Pro 24-bit recording mode and direct SCSI CD-R burning capability (for audio CD's and data backup), the VS-1680 brings your recording space right up to date.

*VS-1680P is supplied with internal 2.1 Gig H/D and one VS8F2 effect board (2 stereo/4 mono FX processors)
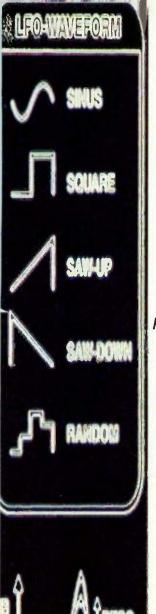
Make it with a Roland

VS-1680 Digital Studio Workstation

Roland

LFO-WAVEFORM

SINUS

SQUARE

SAW-UP

SAW-DOWN

RANDOM

RESO.

CUTOFF-FREQ.

ENVELOPES (EG):

A D S R

Synth School

PART 10: MODELLING ELECTRIC INSTRUMENTS

Last month, PAUL WIFFEN looked at how virtual synthesis can emulate analogue synths whilst going beyond their hardware-based limitations. Now he looks at its applications for imitating and exceeding older instruments such as electric piano and organ.

As I explained last month, virtual synthesis consists, in principle, of recreating what happens inside a 'real world' instrument in the ethereal domain of Digital Signal Processing. The technology involved can be viewed as an extrapolation of effects processing (with which we're all reasonably familiar) back to the point where the sound is first generated. It applies to real mechanical musical instruments as much as to electric or electronic ones where the sound is generated and modified by discrete analogue components.

DRIVER & MODIFIER

When modelling analogue synthesis, software engineers replace each element of the synthesis process (oscillators, filters, envelopes, and so on) with software routines which interact in exactly the same way as their analogue counterparts. The process of modelling earlier musical instruments is actually simpler, in theory, as it separates the process into just two sections, however, the implementation of each of these sections may well be much more complex than the modelling of the individual elements of analogue synthesis). The technical terms for these two sections are 'driver' and 'modifier'.

In the simplest terms, you could think of the driver as what actually produces the sound in the first place — or, to be slightly more scientific, how the energy is initially put into the system. In the

The Technics WSA1; its excellent electric piano models attracted many traditional keyboard players.

case of a guitar, the driver would be the finger or plectrum hitting the string; in a wind instrument, it would be the breath passing through the mouthpiece; in a violin, it would be the bow scraping across the string. These are all actions which produce the initial vibration, and as such they 'drive' the systems.

The modifier is fairly easy to comprehend: it is the part of the musical instrument which takes the initial vibration and changes it into what we recognise as the sound of that instrument. This would be the bridge and sound box on a guitar or violin, the tubing on a wind instrument, and so on.

But before we look at the modelling of traditional western classical orchestral instruments, which are somewhat complex sound-production systems, let's look at how the theory of driver and modifier is applied to some rather simpler electronic instruments which pre-date analogue synthesis.

NEW MODEL PIANO

Although the conventional acoustic piano is such a complex system that an authentic model would cost a fortune in DSP hardware, the somewhat simpler system developed for electric pianos is much more feasible to physically model, and as a result there have been some quite successful models of electric pianos, by Technics on the WSA1 (see the 'Higher Polyphony The Technics Way' box), and Korg on the Z1 (many of which have been bought by traditional keyboard players because of the authenticity of their Rhodes and Wurlitzer patches). The driver in the electric piano system is, of course, the hammer hitting the tine; a physical action. The modifier is the pickup placed over the tine to capture and amplify its sound, and this part of the process is electrical. It may be worth recalling at this point that in instruments referred to as electric (electric guitar or electric piano), the source of the sound is a physical event and the mechanism for amplifying it electrical. In instruments referred to as electronic (such as the organ or synth), the entire sound-generation process is electrical.

Having decided what our driver is, in the case of the electric piano, we have to create a model of what happens when the hammer hits the tine. Clearly, there is a degree of timbral change in the initial sound, based on how hard the key is struck, so not only do we need to vary the volume of the sound but also to create a different harmonic series based on the velocity of the key-strike. The increase in the proportional level of higher harmonics on harder key-strikes is a fairly well documented phenomenon which conforms to the natural increase in brightness which many musical systems exhibit when more energy is put in. This is because higher harmonics require more energy to generate at a given volume (because there are more cycles per second), so when there's less energy present in the system, the amount converted into higher frequencies is reduced disproportionately. This not only explains why a low-velocity key-strike produces a duller sound, but also why the initial strike produces the brightest point in the sound, after which the sound quickly becomes duller. An electric piano sound very quickly approximates to a sine wave at the fundamental frequency of the note. This is fairly standard stuff and will not cause too many problems for any software DSP engineer worth his salt.

Figure 1 shows the parameters for the Electric Piano Model in the Korg Z1. The settings were programmed by producer Martyn Phillips for a Wurlitzer sound. If you look at the Hammer parameters, you'll see that the Wurlitzer is fairly velocity sensitive (76, where 0 equates to no velocity and 99 is incredibly velocity sensitive), but generates very little attack noise. Rhodes patches created with this model tend to have at least a setting of 35 for click, unless they're emulating the DynoMyRhodes electronics, in which case a setting of 75 is more appropriate.

The most interesting part of the electric piano model is the modifier. This is to be expected, as the driver part of the electric piano, the struck tine, is a very small, uninteresting sound (which is why it was easily covered by a sample in the Technics WSA1). The most successful electric pianos used a fair amount of electronic processing to turn this sound into something more interesting to the ear. Clearly, an in-depth analysis of how the sound is modified by such electronics is more the domain of the software engineer creating the model than the musician using the model to recreate his electric piano timbres. Indeed, many of the terms used for the parameters are drawn from electronic circuit
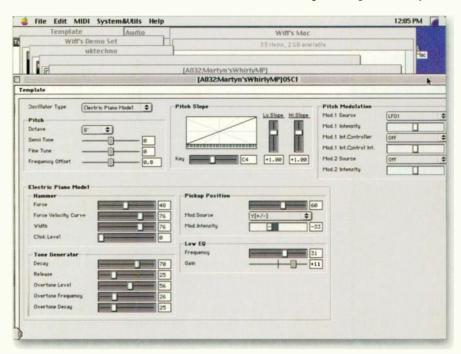
design. However, each separate physical model tends to have one key parameter which leaves you in no doubt about the authenticity of the model (as you'll see next instalment, when I talk about Rosin Amount for bowed string and Embouchure for brass/reed instruments). In the case of Electric Piano models, this key parameter is clearly Pickup Position, which appears in both the Technics WSA1 and Korg Z1 electric piano models.
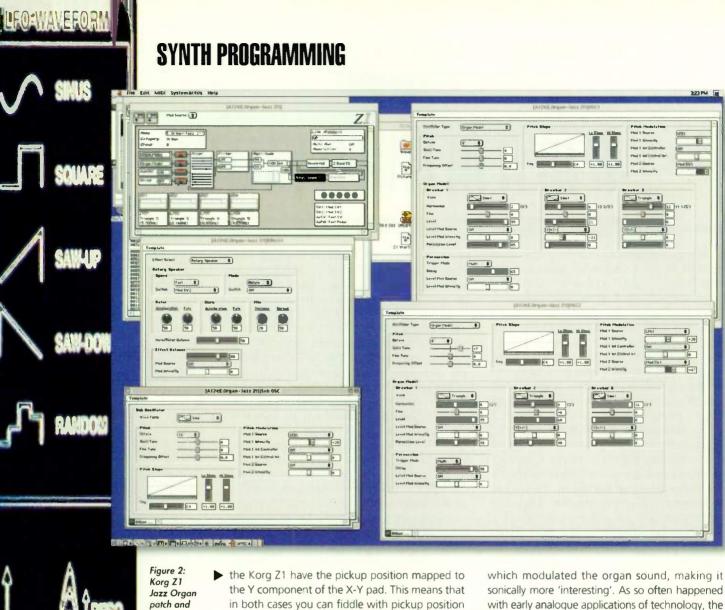
Anyone who owned a Rhodes or Wurlitzer piano in the '70s should remember the fashion for opening them up and individually adjusting the position of the pickup over the tine. It was a very time-consuming process, but was perhaps the best way of customising your sound, as it really did bring about major changes in the timbre of the instrument. At one extreme it was possible to achieve a very bright, thin sound which would cut through anything, while moving the pickup to the other end of its travel yielded a plummy, mellow sound (a bit like the difference between the bridge and neck pickups on an electric guitar).

The joy of physically-modelled electric pianos is not only that this Pickup Position parameter allows you to change the apparent pickup position without all that tedious mucking about inside the instrument with a screwdriver: you can also do it in real time, while you're playing. On the Technics WSA1, pickup position is available on the unsprung mod wheel, while most electric piano patches on ▶

"Technics took an alternative route to giving modelling technology a reasonable amount of polyphony at an affordable price."

Figure 1: Korg Z1 Wurlitzer patch.

SYNTH PROGRAMMING

SINUS

SQUARE

SAW-UP

SAW-DOWN

RANDOM

RESO.

CUTOFF-FREQ.

ENVELOPES (EG):

A D S R

*Figure 2:
Korg Z1
Jazz Organ
patch and
effect
settings.*

▶ the Korg Z1 have the pickup position mapped to the Y component of the X-Y pad. This means that in both cases you can fiddle with pickup position until you get the sound you like and then leave it there (using the X-Y Hold switch on the Z1).

ORGAN TRANSPLANT

Many people are familiar with the fact that the electronic organ works as a sort of primitive additive synthesizer. Drawbars control the level of a series of tone wheels, each of which (in theory, at least) should produce a sine wave representing one of the harmonics in the natural series. These form the driver component of the system, with the rotation of the tone wheels being the original source of the sonic energy in the system. This, of course, dates back to how pipe organs (perhaps the first additive synthesizers) changed the timbre of the sound by adding together pipes of related pitches to create a fuller sound. Electronic organs had as many as 10 drawbars, which gave the ability to mix together the lower pitches in the harmonic series to create different registrations (the latter is originally pipe organ terminology, referring to a series of stops for each of the sets of pipes which were either in or out — ie. on or off). Nowadays, we would probably refer to them as presets, as they essentially change the timbre of the instrument.

This is rather a simplification of what happens inside the most enduring versions of the electronic organ — and we must not, of course, forget the major 'external processor' involved: the Leslie speaker,

which modulated the organ sound, making it sonically more 'interesting'. As so often happened with early analogue applications of technology, the actual product departed from what it should have been according to its 'on paper' design, but was none the worse for that. Indeed, the organs which came closest to producing pure sine waves were the ones often referred to as 'cheesy' these days. The distortion produced in the classic Hammond sound, often a product of ageing tone wheels and abused circuitry, added greater harmonic complexity than simple harmonic addition ever could, often in a similar way to the complex but aurally pleasant distortion produced by guitar amps and distortion pedals. Clearly, a physical model of electronic organs which could only recreate the theoretically pure organ sound would only be of interest to those recreating kitsch '60s lounge music. So organ models need to recreate the more complex phenomena which resulted in the more enduring organ timbres.

The first instrument to use modelling technology to recreate electronic organ sounds was the Technics WSA1. This instrument does not use physical modelling in the purest sense of the term, as the basic source of most driver sounds is samples (see the 'Higher Polyphony The Technics Way' box for a more complete description of Technics 'acoustic modelling' technology). However, for electronic organ sounds, single-cycle waveforms could be added together to model how the basic organ timbre is built up using tone wheels at related frequencies. ▶

SYNTH PROGRAMMING

▶ In Organ mode, the WSA1's backlit LCD display changes to give a representation of drawbars, which can then be modified with the sliders next to them. This means that harmonic content can be changed in real time, just like in all those Keith Emerson solos (although I have yet to see the modulation parameter for routing virtual knives into the cabinet...).

On the Korg Z1, although the assignable knobs below the display can be set to control the level of up to five drawbars (or groups thereof), the way in which organs are modelled is slightly different. Each oscillator can have a different model loaded into it, but the Organ model only has three drawbars (although there are three different variations on a sine wave or triangle wave for each drawbar to control the level of). The best way to make a complex organ sound is therefore to switch both oscillators to the Organ model and then use each one to produce three different

drawbar harmonics. The Sub Osc can also be used to produce the fundamental, so that the six drawbars can be set to higher harmonics. The Jazz Organ patch in Figure 2 demonstrates this very clearly: the Sub Osc is set to the fundamental (16' in classical pipe organ terms), Osc 1 is set to the second, sixth and twelfth harmonics (8', 2⅔' ' and 1⅓') and Osc 2 has two drawbars set to the eighth harmonic (2') and detuned slightly; the third drawbar is doubling up the fundamental.

LESLIE FILLIP

The other section of the Z1 you can see in Figure 2 is brought into play all the time for electric organ sounds: it's the rotary speaker effect algorithm. This gives the Leslie effect, which, as I mentioned earlier, is synonymous with enduring organ timbres. (If you're interested in how the ▶

The Technics KN5000 virtual drawbar acoustic modelling, first used on the WSA synths.

HIGHER POLYPHONY THE TECHNICS WAY

The main obstacle to the development of affordable physical modelling synths over the last five years has been the expense of the DSP hardware. The flood of polyphonic DSP-based machines which hit the market last year (Yamaha AN1x, Roland JP8000, Korg Z1, Nord Lead 2) was very much due to recent decreases in cost and increases in power of DSP chips. Before this price/performance breakthrough, companies working on the development of physical modelling, such as Yamaha and Korg, were forced to limit the polyphony of their early instruments (the VL1 and the Prophecy) to a single voice. This wasn't too much of a problem, as these instruments were principally designed to recreate the voicing of monophonic instruments such as brass and woodwind or lead and bass synths, and it allowed all the power available to be concentrated into a single powerful voice.

However, Technics took an alternative route to giving modelling technology a reasonable amount of polyphony at an affordable price. They realised that the greatest amount of DSP power was taken up by producing the driver, the original sound before the resonator modifies the harmonic content. By replacing a modelled (and therefore processor-intensive) driver with a PCM sample, they could save an enormous amount of processing power

power which could be ploughed back into increasing polyphony. As a result, when other modelling synths on the market were duophonic at best, the WSA1 had 32 voices of polyphony, a figure still not achieved by the most powerful current modelling synths. (With the optional 6-voice expansion, the Korg Z1 still only clocks in at 18 voices, for about the same retail price that the WSA1 had on its release in 1995.)

To make PCM samples work as drivers for a modelling synth (rather than as the more 'finished' sound you'd normally expect from the PCM sound sources in an S+S synth), Technics had to record the samples in as primitive a way as possible, removing as much of the resonator component as they could from the samples. So strings were miked as close to the string and as far away from the sound hole of violin or guitar as possible, while woodwind reeds were sampled without the resonating column component. This makes the raw samples in the WSA1 rather less polished and exciting than those in the average PCM-based synth. A perfect example is the source samples for the electric pianos, which have much of the toy musical box about them when heard unmodified.

Fortunately, no-one is expected to listen to these raw samples as they were recorded. When the sampled driver is

passed through the DSP resonator section, the acoustic modelling process recreates the same timbral and enveloping changes which take place inside the instrument once the initial sound has been triggered. So what's the advantage in this? Why not just use a sample which has the final sound of the instrument?

The answer to this lies in the increased expressivity which can be achieved by modifying the resonating component with real-time controls. The amount that can be done with 'finished' samples is fairly limited, especially with fairly crude 'analogue'-style filters and envelopes, which can make the sound brighter or duller, and end sooner or later, but can't make it fundamentally different. The harmonic variation which modelled modifiers can introduce is much more akin to the kind of filtering offered by Emu's Z-plane synthesis, as represented by Morpheus and UltraProteus, for example. Modifications are at the same sort of level as multi-band graphic EQs with serious amounts of cut and boost available, or even the more complicated parametrics which can precisely tailor frequency components.

The much more complex and subtle variations which can be produced by modelling the modifier component of an instrument get to the heart and soul of an instrument's expressivity, without the problems that cross-switching or cross-fading samples brings. In the case of the electric pianos on the WSA1, you can control parameters such as pickup position and pickup sensitivity, as well as the timbral effects of the electronic circuitry itself. Using DSP for the modifier side of the process meant that much of the subtlety and precision of physical modelling could be introduced without the cost of the drivers being produced entirely by DSP.

Sadly, the WSA1 (and its module counterpart, the WSA1M) were not major hits in the S&S-dominated market. They
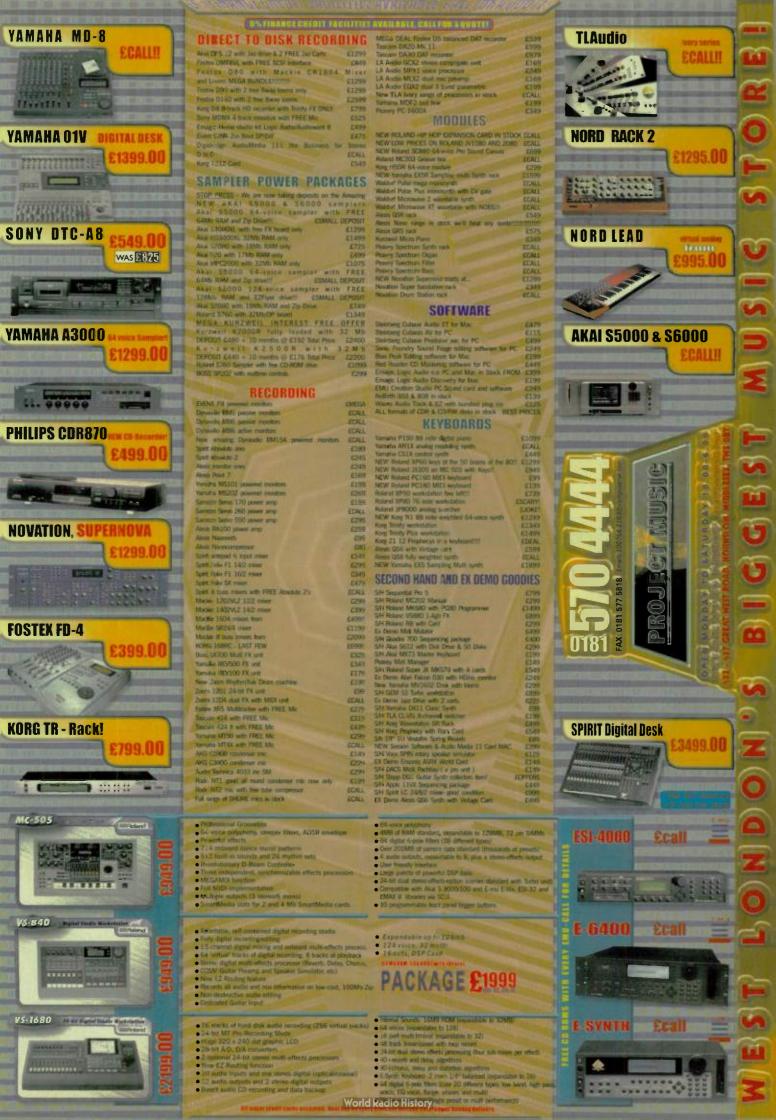
were launched at a time when the demand for subtlety of expression for players and authenticity of real sounds was an all-time low, and the market requirement was for sounds suited to techno and related dance styles. Having discovered some potential for the type of real-time modulation favoured by the dance producers in Technics' acoustic modelling process, this author participated in the production of an expansion board full of drum loops and analogue oscillator timbres, in an attempt to save this developing technology from becoming obsolete before it matured. Unfortunately this Dance board was released too late to save the instrument from the ignominy of the discounted blow-out, and the last WSAs were sold off at a quarter of the original RRP, complete with the Dance board. However, anyone who picked one up at this final price (or buys second-hand at a similar price) secured an amazing deal, as the value for money of the original RRP has still to be equalled in terms of polyphony, if not fidelity, by the modelling synths of the present day.

Technics chose not to continue the technology of acoustic modelling in the form of follow-up WSA products, but one component at least lives on: the way in which the WSA's electronic organ sounds was programmed has been used in Technics' KN5000, and it's from this instrument that we've sourced the LCD screen illustrating the modelling of organs via virtual drawbars.

The Technics WSA1R rock modelling synth.

World Radio History

LFO-WAVEFORM
SINUS
SQUARE
SAW-UP
SAW-DOWN
RANDOM

SYNTH PROGRAMMING
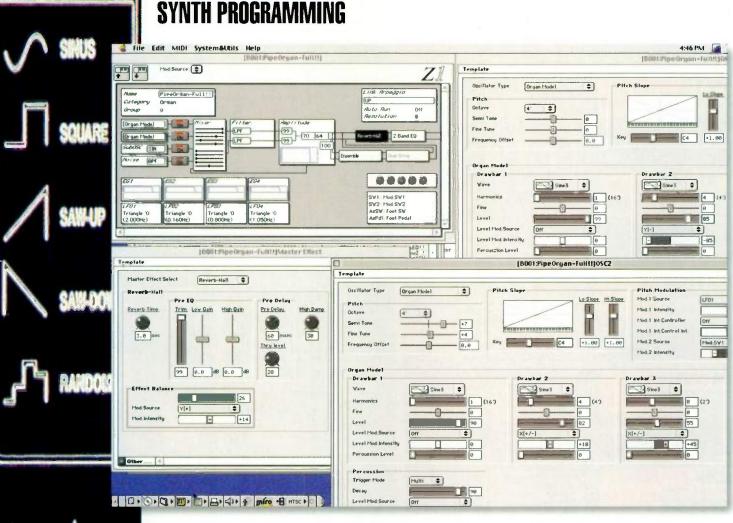
Figure 3: Korg Z1 Pipe Organ patch and effect settings.

▶ Leslie cabinet works, or the history of Hammond organs, take a look at our 'Vital Organ' feature in the October 1997 issue of *SOS*). Indeed, where such organ sounds are concerned, the Leslie effect is the major part of the modifier, in that (apart from some distortion caused by knackered circuitry, key-clicks caused by worn contacts, and so on) it is the rotary effect which gives the sound its character and charm. This is where the line between physical modelling and DSP effects blurs to the point where one can be seen as part of the other. In fact, a physical modelling instrument which could not apply a rotary speaker effect could hardly be said to properly cover organ modelling. Fortunately, both the Z1 and WSA1 (the only two synths which claim to cover organ modelling) both have effects algorithms for rotary speaker simulation.

As you can see from Figure 2, the proper modelling of a Leslie speaker includes parameters for the rate and acceleration of both the rotor and the horn, as well as for the distance and spread of the virtual microphone which is picking up the sound. Mod switch 2, just next to the X-Y pad on the Z1, is normally used to swap between the slow and fast rotation rates.

The Organ models on both the WSA1 and Z1 are not just restricted to the modelling of electronic instruments. Both are extremely adept at pipe organs of the ecclesiastical variety, although in both cases the rotary speaker is best eschewed in favour of the largest hall reverb available on the machine. Figure 3 shows a typical Classical Pipe Organ patch.

You will notice that all modulations have been switched off and that the click component, so common in electronic organs, is also defeated. Then it simply remains to select the required footages (remembering, again, that the Sub Osc can be used to add in an extra footage at the bottom end) and give the hall reverb its largest possible setting.

ORCHESTRAL MANOEUVRES

This move from the electronic to the acoustic world (albeit still within the digital domain) leads rather nicely into the remaining chapter on physical modelling, coming your way next month. I'll be looking at plucked string algorithms (which produce both acoustic and electric guitars, harpsichords, and other plucked string instruments, such as dulcimers and spinets), and at the three most widespread uses of physical modelling in the acoustic world — brass, reeds and bowed strings.

Although the balance of this piece has been based around Korg's MOSS system (with a small contribution from Technics' acoustic modelling), next time I'll be broadening the palette to include the Yamaha VL system in its many incarnations, including the cheapest physical modelling unit to date, the VL70M. Until then, see if you can lay your hands on a WSA1 or Z1 to try out some of the electric piano and organ sounds we've been looking at. If not, a Korg Prophecy will be good preparation for next month, as it also covers plucked strings, brass and reeds. **SOS**

Millennium

EMU AUDITY 2000 DIGITAL SYNTH MODULE

Modular?

The streamlined '90s exterior of Emu's Audity 2000 claims to hide a virtual modular hybrid, harking back to the company's own lost '70s analogue megasynth. DEREK JOHNSON wonders how they plan to cram it all in...

Emu's Audity was a late-'70s one-off: an over-specified microprocessor-controlled monster analogue synth years ahead of its time. Based on 16 separate synth cards that were voiced and controlled by a software front end, the Audity offered glimpses of what was to come in the future. Had it gone into production, the Audity would have retailed for something like US$70,000. But it was not to be: a pre-Audity prototype was commissioned by, and built for, ex-Tangerine Dreamer Peter Baumann, and one Audity was constructed (now owned by David Kean of the Mellotron Archives, www.mellotron.com). Emu subsequently moved into sampling, with the first Emulator, and the rest, as they say, is history.

The intervening years have seen Emu develop a fine line of samplers under the Emulator and Emax names, with an enviable in-house sound library, which was repackaged during the late '80s in various upmarket rackmount sound modules. Of these, the Proteus family was a hit, and survived in one form or another until quite recently. Even post-Proteus Emu synths have a similar basic architecture and user interface. Newer modules have seen Emu move sideways; rather than repackaging more of the Emulator sound library (along with admittedly clever synth sections), the company have tested the water with trend-setting — or perhaps trend-following — modules aimed at contemporary musicians. Carnaval and Planet Phatt (reviewed in *SOS* July '97 and June '97 respectively) are both the result of this course of action, while the earlier Morpheus was an attempt to create something different, with morphing Z-Plane filters (reviewed December '93; Morpheus FX, May '94). Emu's latest synth, however, takes a chunk of Emu pre-history and drags it into the late '90s, and perhaps beyond.

INTRODUCING THE AUDITY 2000

Emu's advertising literature identifies the Audity 2000 as a 'Digital Rhythmic Modular Synthesizer'. The term 'Rhythmic' refers to its multiple arpeggiators and the way in which incoming MIDI clock can be used to sync LFOs and envelope generators (it doesn't have the rhythmic Beats mode found on recent modules such as Carnaval and Planet Phatt). 'Modular' in this case refers to a comprehensive modulation matrix, where virtual patch cords are used to interconnect various modulation sources and destinations, extending the instrument's synthesis capabilities even further. You might have expected Emu to jump on the analogue-modelling bandwagon with the Audity 2000, especially in view of this 'modular' patching metaphor, but they've stuck with the tried and tested S&S (Sample & Synthesis) sound-generation method, with all the benefits that brings for polyphony and multitimbrality (see Paul Wiffen's explanation in this month's Synth School, starting on page 168, of why modelled instruments have such modest polyphony and multitimbrality).

While sharing the 32-voice polyphonic, 16-part multitimbral architecture of previous Emu modules, the Audity 2000 differs in a number of significant ways. For a start, the 16Mb of basic sampled waveform ROM is aggressively synth-orientated — there isn't much chance of emulating a General MIDI module here! And if that's not enough, an internal slot allows another 16Mb of waveforms to be added with an optional card. The Audity 2000's filtering has been given a lot of thought, and is derived from the Z-Plane models developed for the Morpheus. Another nice touch is the inclusion of two effects processors (albeit simple ones). Not every previous Emu module is so equipped.

The Audity's most significant features, though, have to be its virtual 'patch cord'-based modulation matrix, and its arpeggiator. Wait... did I say "arpeggiator"? I actually meant 16 arpeggiators! Yes, the Audity has one arpeggiator for each of its 16 parts, with full user editability and MIDI sync!

The new synth also has a different look from previous Emu modules; it's not soberly grey as were the original Proteus family, and it's not as self-consciously jazzy as the Carnaval Latin module. Its blue front panel is completely different from what has come before from Emu: the stylish cutout by the display is flanked by a total of nine buttons, plus five small knobs and one big one. The 2-line x 24-character display is all that's similar to previous Emu modules. Four of the smaller knobs (one is a volume control) provide real-time control over assignable parameters, and a Shift button cycles through two sets of parameters and four arpeggiator parameters.

Of the nine buttons, one is the Shift button mentioned above, two are arrow keys which move between parameters in a window, one offers a Save/Copy option, and another is an Enter button. The remaining four work as follows:
- Edit: press this to go into edit mode.
- Clock: allows you to select the Audity's clock source for the arpeggiator and anything else that's clockable on the Audity, such as the LFOs. Tempo has a range of 1-300bpm, which I think you'll agree is quite sufficient!
- Master: offers a menu of global settings, such as the Master effects, Master arpeggiator, Transpose value for the whole machine, sundry MIDI operation parameters, and so on.
- Arpeggiator: this button takes you to a menu where you can select whether arpeggios are sent over MIDI, edit user arpeggiator Patterns, and define master arpeggiator settings.

The rear panel hosts three pairs of audio outs, two sets of which can be used as effects send/return loops for Audity voices, or as auxiliary ins if you need extra mixer inputs. One major surprise is the Audity 2000's digital output, which offers a choice of S/PDIF or AES/EBU, for ease of interfacing to, say, a digital desk. The power supply is internal, and automatically senses what sort of mains supply it's plugged into.

SOUND ARCHITECTURE

The Audity's 16Mb waveform ROM is divided into 287 raw waveforms. All the traditional synth sounds are here — sine, sawtooth, square wave — together with sampled synth basses, hits, percussion, a couple of drum kits and some plain weird noise. There are several instances where waveforms get a little repetitive, but on the whole it's an excellent collection.

Audity patches — or Presets, as Emu call them — are made up of four elements, dubbed Instruments (earlier Emu modules had two elements per voice). Each Instrument is assigned a waveform, which is followed by a fairly traditional representation of an analogue-style signal path; in fact, each Instrument is virtually a full synth in its own right, and can be split or layered within a Preset. Presets can also be 'linked' to one or two other Presets. This is the closest you get to a 'performance' mode with an Emu module, and allows Presets to be easily split or layered with each other; the result functions as if it were one Preset. Obviously, the more Instruments you use per Preset, and the more Presets you link, the less overall polyphony you'll have available.

The factory selection of Presets is quite varied, offering leads, simple waveforms, complex, harmonically rich sounds, tempo-based, LFO-driven Presets, keyboard emulations, hits, organs, edgy, cutting sounds, basses, noises, pads, vocals, bells, effects percussion, and scratches. There are seven banks of Presets, totalling 896 altogether; banks 2-6 contain ROM factory sounds, while banks 0 and 1 are are available for users to store their own Presets.

INVESTIGATING INSTRUMENTS

OK, so each Preset is made up of four Instruments; what is each Instrument made up of? In the order they arrive while you're scrolling through the display, here are each Instrument's parameters:
- Waveform.
- Key and velocity ranges, for splitting or layering, with crossfading.
- Real-time crossfade, which is controlled by a patch cord in the modulation matrix.
- Transpose, +/-36 semitones.
- Tuning: +/-36 semitones coarse, and fine-tuning in 1/64th semitone steps.
- Amplifier Volume (-96dB to +10dB) and pan position.
- Volume Envelope: choose from the factory envelope that goes with a given waveform, a standard, time-based, six-stage envelope (each with its own rate and level value) or a tempo-based option which lets you set up envelopes that respond exactly in tempo with incoming MIDI clock.
- Chorus: this is in addition to the onboard effects, and is a simple device, with just two parameters (amount and width). Using this effect requires two voices of polyphony.
- Sound-Start offset: this parameter lets you start a sound from elsewhere in a waveform than its beginning, while the Sound Delay parameter defines the time between when you hit a note and when the current Instrument actually plays; there are tempo-based options for this feature.
- Non-Transpose: fixes the pitch of a given Instrument. You can't specify what pitch this will be, so this feature is perhaps best reserved for noise elements of Presets — transient attacks and so on.
- Solo Mode: turns a given Instrument into a monophonic voice; various triggering options (low, high, last note, etc) are available, so that you can emulate classic monosynths of the past.
- Assign Group: allows you to manage your 32 notes of polyphony and avoid note stealing.
- Glide: a portamento effect, with editable rate and three curve options.
- Filter: a Z-Plane modelling filter, as found on the Morpheus. This is one of the most flexible and powerful that money can buy. It can be low-pass, high-pass, band-pass, notch, parametric, and much more (50 types in all), and can change dynamically over time. Look

" ... did I say "arpeggiator"? I actually meant 16 arpeggiators! Yes, the Audity has one arpeggiator for each of its 16 parts, with full user editability and MIDI sync!"

EMU AUDITY 2000

▶ out for the vowel formant options in particular.
- Filter Envelope: has the same features as the volume envelope, except that negative level amounts are available for each stage. Applies an envelope to the filter frequency, for filter sweeps.
- Auxiliary Envelope: an extra envelope with the same facilities as the filter envelope, which

can be freely patched where you need it.
- LFO 1 and LFO 2: identical, and sync'able to MIDI clock or the master tempo (great for gated effects, for example). Waveforms include triangle, sawtooth, sine, square and various pulses, and there's a bonus 'variation' parameter, a random effect that sounds something like chorus in use.
- Patch Cord assign: this is where you work with the modulation matrix — more in a moment.
- Pitch-Bend range: up to 12 semitones.
- Mix Output assign: you can choose from the main or one of two sub outs, and use hard panning to assign Instruments to one output only.

Each Preset also has a number of parameters that affect all four Instruments at once. The two effects processors, for example, aren't Instrument-specific, although you do have a choice of using Master effect settings or programming effects specifically for the Preset. The arpeggiator is also set globally for the whole Preset. Other parameters give you a choice of alternate tunings (the small selection includes gamelan and 19-tone, plus one user-definable tuning) and provide control over links to other Presets.

So far, I haven't really mentioned drum sounds: well, there are a handful of drum-kit waveforms, but the user has no control over which sounds go where in a kit, or over parameters on a per-drum basis. The drum kits themselves, however, are a good selection, with both electronic and acoustic examples.

PATCH CORDS

Central to getting the most out of the Audity 2000 is the concept of Patch Cords; a modulation routing system where various sources are patched to various destinations — there are 64 of each. Sources, which can include MIDI controllers as well as internal

parameters, are connected to destinations by virtual Patch Cords, and there are 24 Cords for each Instrument in a Preset. The Patch Cord-related parameters provide an easy way to modulate parameters with unusual control sources, just as if you had a modular synth; some of the control sources can themselves be modulated, so pretty complex results can be produced.

Hands-on modulation is provided by the four control knobs on the front panel, which can be shifted to provide eight modulation sources, as mentioned earlier. Each knob responds to and transmits a MIDI continuous controller, and, of course, pitch-bend, mod wheel and other data transmitted from your master keyboard can be used as modulation sources. Modulation destinations are numerous, some obvious (pitch, glide, filter frequency and resonance) and some less so — interesting results can be obtained, for example, by modulating the various stages of the EGs while a sound is playing. Esoteric sources and destinations really need experimentation, and are not explained well in the manual. I would direct you immediately to the Patch Cord tutorials, which offer some very enlightening examples, and encourage the necessary experimentation. Interestingly, the Patch Cords themselves can each be a modulation destination!

As with a true modular synth, you simply keep patching sources and destinations together until you get an interesting result. The difference with the Audity 2000 is that you can save the patch in a user memory! As intuitive as Emu's user interface is, the Audity 2000 would really benefit from a graphic editor, especially for dealing with Patch Cords; I really want to see what I'm patching where, rather than visualising it. Emu could have taken a leaf out of Clavia's book here; their Nord Modular was designed to be patched via its PC-based graphic front end. I await the necessary Audity 2000 templates for generic synth-editing software with interest.

ARPEGGIATOR

Patch Cords might make for intriguing synthesis possibilities, but it's the arpeggiator (or arpeggiators, rather) that will make the most immediate impact on potential Audity 2000 owners. Indeed, Emu actually place the chapter on the arpeggiator early in the manual, before talking about voice editing and so on, which shows you how important they think it is. As mentioned earlier, each Preset can have its own arpeggiator (so you could have 16 Presets all chugging along with their own arpeggiation), or it can use the Master arpeggiator. Indeed, the Master arpeggiator can be used by all currently selected Presets, if you like.

Standard arpeggiations are available — up, down, up/down, forward assign, backward assign (these both play back notes in the order in which they were pressed), forward/backward, random — or you can choose a Pattern, which stretches the concept of the arpeggiator rather a lot. There are three banks of 100 Patterns on board, and Bank 0 ▶

EFFECTS

While it's nice to see effects on the Audity, don't get too excited: they're basic and serviceable rather than comprehensive. That said, they're easy to use and sound good, and Emu have provided a number of useful routing options. The Audity basically has two effects processors:

- **FXA:** this group consists of 44 reverb and delay effects — various rooms, halls, and so on — with control over decay and HF damping, and an FXB-to-FXA routing parameter.
- **FXB:** this group offers 32 choruses, flanges, ensembles, some delays and distortion effects, with control over feedback, LFO Rate and Delay.

As with the arpeggiator, there are Preset-specific and Master effects. The Master effects would be ideal for use multitimbrally, and it's also possible to use the effects from a Preset as Master effects.

GETTING SERIOUS!

With Shure BG Series microphones your future's sounding great. No one gives you more pro-quality sound for your money than Shure BG. Six high-performance, highly durable microphones each engineered to the toughest standards in the business. If you're serious about going places, make sure your microphones are up to the job. BG microphones from Shure will show them how serious you are.

BG1.1 DYNAMIC MICROPHONE

BG3.1 HIGH-OUTPUT DYNAMIC VOCAL MICROPHONE

BG4.1 HIGH-OUTPUT CONDENSER INSTRUMENT MICROPHONE

BG2.1 HIGH-OUTPUT DYNAMIC VOCAL MICROPHONE

BG5.1 HIGH-OUTPUT CONDENSER VOCAL MICROPHONE

BG6.1 CARDIOID DYNAMIC INSTRUMENT MICROPHONE

SHURE®
BG microphones

FOR THE FULL STORY ON SHURE BG MICROPHONES AND INFORMATION ON OTHER QUALITY
PRO AUDIO EQUIPMENT FROM HW INTERNATIONAL, CALL OUR FREEPHONE TODAY!

FREEPHONE TO RECEIVE
OUR PRO AUDIO
CATALOGUE NOW!
0800 7311990

> "This is a sophisticated, flexible and good sounding synth that will suit die-hard programmers down to the ground."

▶ is for user Patterns. These can be up to 16 steps long, and you're free to define note value offsets (+/-48 semitones), velocity value, and note length. A Repeat parameter repeats each step up to 32 times.

Setting up the arpeggiators is relatively simple. First, select a mode — one of the standard arpeggiations or a Pattern — and a note value (anything between a double whole note and a 32nd note, with triplets and dotted values available). You can then alter the 'pattern speed', which doesn't effect tempo but multiplies or divides the selected Pattern's note values, so that the Pattern just *seems* to play back faster or slower. Values can be between a quarter and four times the incoming tempo. You then choose a velocity value for the arpeggiated notes, and a gate time, which sets how long the notes will be.

The Extension Count and Extension Interval parameters define a regular transposition of the current arpeggio. For example, if you set the Extension Interval to two semitones (the maximum is 16), and the Extension Count to one (the maximum is 15), your chosen arpeggio will play once with the keys you've held down, and again transposed up two semitones, before repeating. The onset of an arpeggiation can also be delayed, and you can even define the length of time an arpeggiation will play for. A Recycle parameter works with delay and duration, resetting the pattern to the beginning when the duration value has been reached. Usefully, there's also a Latch facility: just touch a few notes or a chord and they'll keep arpeggiating until you play another chord if Latch is activated. Lastly, a key range keeps the arpeggiations within a user-defined section of the keyboard, so you can have mad arpeggiations under one hand, while soloing with the other. The factory Patterns include options aimed at drum & bass merchants, hip-hoppers, ravers, and all sorts, many of them very good. There is, however, no list in the manual!

IN USE

Using the Audity 2000 is rather like having a hybrid of a perfectly normal S&S instrument and the modulation matrix from a modular synth. With Korg's classic MS20 patchable monosynth, you could create perfectly serviceable sounds by tweaking the knobs alone, but life got really interesting when you used patch leads to break into the signal path and add weird modulations. The Audity 2000 has a similar feel.

Emu synths have always been easy to use, and the Audity 2000 is no exception. Many may find such a small display to be a hindrance, especially in the late '90s, but Emu's operating system is so streamlined and straightforward that it's not really a problem; I'd still like graphical editing software for the Patch Cords, though.

The arpeggiator is nothing short of brilliant; where else would you get 16 of the blighters side by side? Some of the preset Patterns are excellent, and none are less than valid for the style of music being emulated. Also worthy are the real-time control knobs, although I'm not sure how

accessible they'll be once the Audity is placed in the average keyboard player's rack.

On the subject of sound quality, the Audity 2000 is a winner; the raw waveforms are of a high quality, and reasonably varied, though there are too many one-shot percussion choices — 81. They are very effective, but many sound too similar; the same goes for the 52 looped percussion samples and sundry one-shot scratches. I also found too much similarity across the factory Presets, although a more upfront, big-sounding collection of sounds, both abstract and musical, you'll be hard-pressed to find. I think the emphasis has been on the abstract, rather than the immediately musically useful, but as examples for your own work, the factory Presets are excellent.

CONCLUSION

This is a sophisticated, flexible and good sounding synth that will suit die-hard programmers down to the ground. Should they get their cheque book out? Well, the £1300 price tag may cause them to falter slightly. It has to be said that a couple of full workstation synths are within shouting distance of that price, and that you'd get £200 change from the asking price of the Audity 2000 when buying an Akai S3000XL sampler.

However, there are things about Emu synth modules which go some way towards justifying their higher price tags. The company provide good build quality, excellent raw sonic material, and truly interesting sound-creation facilities with an easy-to-comprehend front end. And, to their credit, they don't turn over their range every six months or release cheaper versions of existing products with annoying regularity. On the value-added side of the equation, the Audity provides a superb synth engine with a range of well-implemented traditional facilities, plus an excellent modulation system to keep the dedicated programmer happy for ages. And that's what it's all about: sonic potential. This is why classic gear has gone through an inflationary *renaissance* over the last decade. Other balancing features include the well-specified arpeggiator — how much would 16 of those cost, if you could actually find them independently of other synths? And perhaps another manufacturer might have left off the digital output, offering it instead as a £300 upgrade. Finally, let's not forget that potential $70,000 price tag of nearly 20 years ago for the original Audity! Things have come a long way; Emu have pushed the envelope of S&S with the Audity 2000, and it's a worthy successor to their original grand concept. **SOS**

" Eugen Beyer " founder of beyerdynamic, GmbH & Co.
shown testing the World's First moving coil Headphones.

A Bright Past...
A Brilliant Future.

DT 48

Audio history. In 1937, beyerdynamic introduces the world's first moving coil stereo headphone, the legendary DT 48.

Audio future. In 1998, beyerdynamic introduces the world's first digital recording microphone, the MCD 100.

Established in 1924, beyerdynamic's legacy of innovation, fidelity and reliability is recognized and honored by audio professionals around the world. 73 years of audio excellence.

An ear to the past, an eye to the future.

beyerdynamic... *Fidelity in Audio.*™

MCD 100

beyerdynamic)))))

for more information

Tel. 0800 374994   e-mail: sales@beyerdynamic.co.uk   Burgess Hill, RH15 9TN

Open up.
And say ahhh...

Then say hello to Roland's JV-2080 Synthesizer Module. It's the latest evolution in the world's most popular line of sound modules.

With 8 expansion slots, the JV-2080 sets a new standard for user-expandability, giving you access to more than 2,900 patches when fully expanded. To help you take advantage of all these classic Roland sounds,

The large LCD and Patch Finder make it easy to locate and audition any patch in seconds

we've added the Patch Finder so you can hear and audition only strings when you need strings, separate your basses from your brasses, and quickly locate any of your favourite internal or expansion patches.

Once you've selected those perfect patches, the JV-2080's 64 voices, 3

Use 3 insert effects simultaneously, each with 40 effects selections in addition to reverb and chorus.

simultaneous insert effects and 3 stereo outputs provide a flexible production environment that just may leave you speechless. So check out the JV-2080 at your local Roland dealer. And now there are 2 new Expansion Boards to add to your collection: Techno and Hip-Hop. Call 01792 515020 now for your Free DEMO CD, featuring this and many more exciting products from Roland.

Chances are, you won't stay silent long.

Make it with a Roland

MAKING MUSIC AWARDS
Best Module
Roland JV-2080

Roland JV-2080 64 VOICE SYNTHESIZER MODULE
3x EFX 8x EXPANSION

USER:001 EXPAND!

FREE DEMONSTRATION CD

Roland®

Shelf Help

REORGANISING YOUR STUDIO ON A BUDGET

Getting your studio organised doesn't necessarily mean investing in expensive studio furniture or being a DIY whizz. DEREK JOHNSON & DEBBIE POYSER explain how they made the most of their space, and pass on some hints and tips for a tidier life.

For years, our recording gear was arranged around our studio room on a motley collection of furniture, mostly consisting of various cast-offs from other rooms in the house. It was generally rather untidy and sometimes inconvenient. Then, a few years ago, we saw an article written by Mike Simmons about Studio Ergonomics (see *SOS* September 1993) in which he described the way he had laid out the small studio area under the staircase in his living room. He'd used standard Spur shelving to some extent, and his main work surface was simply a piece of worktop supported at either end by small filing cabinets. We were taken by how neat and compact it all looked, and when we eventually got around to sorting out our own studio environment, we thought we'd use a similar approach. However, we didn't have two small filing cabinets on which to support a work surface, and it seemed perverse to buy some to hold up what's essentially just a big shelf! So we decided to see if we could do the whole thing with Spur shelving. Like most non-professional musicians, we didn't have the means or the inclination to turn our gear/playing room into the equivalent of Abbey Road Studio 1: we just wanted a system that would be tidy, efficient and comfortable to use, would look businesslike, and didn't alter the fabric of the room too much (if you ever want to sell your house, this may be a consideration: can you turn that spare bedroom back into a spare bedroom fairly easily? It isn't every potential purchaser that wants an 8-track programming suite/studio in their new abode).

STAYING ON THE SHELF

Spur shelving probably isn't the absolutely cheapest way to construct a built-in studio area, but it certainly is versatile (you can change the shelf configuration whenever you want), and since it dismantles so easily, you can take it with you if you move house. Also, if you're not some kind of DIY genius (we're not) it's super-easy to conceptualise and put up. If you're not familiar with this shelving system, it consists of metal battens, called uprights and
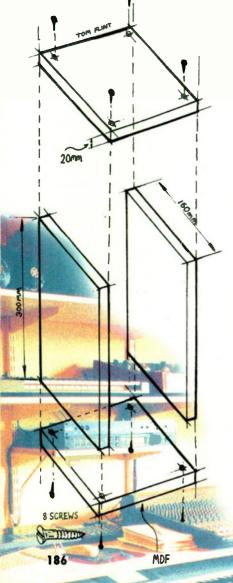
available in different lengths, which are U-shaped in section and feature pre-drilled pairs of slots at regular intervals along their length. These are screwed into your wall (how many you need depends on how big an area of wall you need to span) and matching metal brackets are slotted in to support your shelves. Shelves can be solid wood or melamine-covered chipboard, but the amount of weight the shelves will carry varies according to the shelf material you use (see 'Shelving Statistics' box). Almost all DIY stores stock this type of shelving system, though they vary slightly from manufacturer to manufacturer. They're usually also available in different colours (most often white, cream and black). If you match your shelves to the metal supporting parts, it can look really smart. An all-white system will feel spacious and light, while an all-black system would look suitably hi-tech.
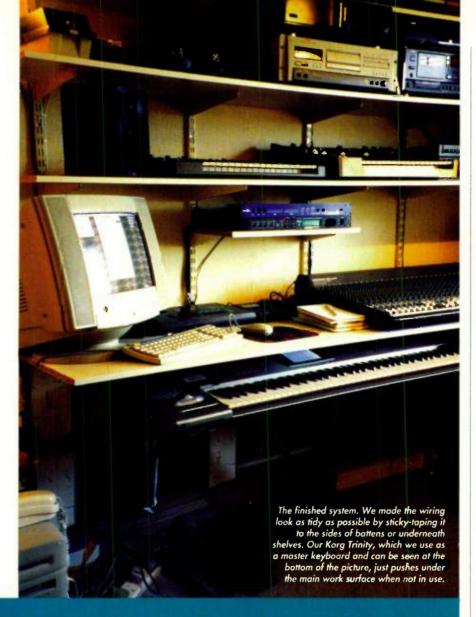
WEIGHTS AND MEASURES

We had a flat expanse of wall, about nine feet across, and we planned that this would host much of our gear. We wanted a main work surface, at desk height, and sufficiently deep for comfort: this would support a computer monitor and keyboard, a mixer, and an 8-track open-reel tape recorder, plus any other little bits that needed to be squeezed in. Above this would be as many shelves as we could fit in, all of which would run across the full width of the wall. These didn't need to be quite as deep as the main work surface, but we had to make sure they would be deep enough for gear like a DAT machine, cassette decks, small analogue monosynths, and so on. Nearfield monitors would also go on one of these shelves. Towards the top of the wall, the shelves could be a lot narrower, as we were planning to use the top couple for books, ▶

TOM PLINT

20mm

150mm

300mm

8 SCREWS

MDF

The finished system. We made the wiring look as tidy as possible by sticky-taping it to the sides of battens or underneath shelves. Our Korg Trinity, which we use as a master keyboard and can be seen at the bottom of the picture, just pushes under the main work surface when not in use.

the time, it works wonders for your memory. We found one for about £5 in an office supply shop, and it's half dry-wipe and half cork-board, so we can pin little notes on it too.

• Make your patchbays look professional (and easier to read) by setting up a template in any graphics package so that you can type the routing labels instead of hand-writing them in those titchy little spaces designed for leprechauns to write in. This is very easy to do if you have a computer and printer. Alternatively, Studiospares can sell you a pack of sticky labels produced by patchbay manufacturers P&R Audio and pre-printed with a selection of useful legends. You get two sheets for a mere £1.99

• If you set up a Spur shelving system and find you have a teeny bit of space left at the top of your wall when you've calculated the optimum spacing of your shelves, put in a very narrow shelf and use it for storing a row of DAT or cassette tapes. You might then find you need a stool to stand on to reach the tapes (the classic Kick-step takes up very little space and can also be used as extra seating in the studio).

• According to our last reader survey, about half of you out there own a guitar. If you have one, or especially if you have more than one, it saves floor space if you buy a couple of guitar hangers and hang the guitars on a wall. It also saves you from falling over them all the time.

• If you've decided to go ahead and re-design your studio layout, you're going to have to embrace the painful reality of unplugging all your carefully constructed wiring and taking apart your existing setup. While you're at it, do think about whether

anything else needs doing in the room. If you need a new carpet, say, or the skirting boards could do with a lick of paint, get it sorted now, if you can, while everything is awry anyway.

• Once you've got everything set up and looking neat, a couple of things will help you keep it that way. Some items of gear work a lot better in the long term if they're kept dust-free (notably mixers) and a few fitted dust covers wouldn't go amiss. There are commercial manufacturers (see 'Manufacturers & Suppliers' box) that make these. Failing that, a mini vaccum cleaner with a tiny nozzle can be very useful; Maplin sell one for under £15. Or watch out in those 'Hi-Tech Tat' catalogues that periodically fall out of your *Radio Times* for tiny nozzle attachments that go on ordinary vaccuum-cleaner hoses.

You could use an ordinary wooden coat-hook bar to store some spare leads on the back of a door.

You can put little 'shelfettes', like the one here, anywhere you like between the upright battens, to make best use of the space between the full-length shelves. If you think about it in advance, you can make small shelfettes the right size to accommodate 19-inch rack units.

disk boxes, open-reel tape boxes, and so on. And why let all that space under the main work surface go to waste? Like Mike Simmons, we decided to put in two long narrow shelves, which would also house books without getting in anyone's way.

The next step was to check on the commonly available sizes of uprights and brackets (the deeper the shelf you use, the longer the bracket you need), and lengths and depths of ready-made shelving board. We found that the longest length usually available, approximately eight feet, was just about right for our wall, and the shelves were available in lots of different depths, including a two-foot depth that was perfect for a main work surface. The manufacturer's recommendations dictated that we use five uprights to span our wall, and when you want the resulting shelves to take a lot of weight it's not worth skimping in this area. Likewise, with brackets, we used ones that were almost as long as our shelves were deep, just for safety. One thing that's very much worth knowing is that you don't have to buy the fixings packs made by the manufacturers of the shelving systems. These are invariably just a handful of overpriced screws and wall plugs, and if you examine them and make a note of their sizes, you can then find the hardware area of the shop you're in and get bags containing hundreds of the same-sized bog-standard screws and wall-plugs for the same price as a dozen or so of the 'special' ones.

PLAN AHEAD

When you assemble Spur shelving, the nature of the design means that a small gap, of around an inch to an inch and a half, is left between the back of the shelf and the wall. This is perfect for snaking cables behind to connect up your gear, but if you do this, be aware that any items with captive mains plugs will have to have their mains plugs removed to let you pass the cable behind the shelf, then re-attached (or wait until everything is in place before you fix the shelves to their supports). You'll also have to take off the plug again if you want to move the item in question too far from its original position. We perched a couple of multi-way mains outlet sockets on the upper shelves (with the leads

going behind the shelves) to accommodate gear that needed to be higher up — this was both for tidiness and because few mains leads would be long enough to reach the floor from a height of a little over 1.5 metres. Plan carefully when you begin to spread out the gear on the shelving system, to avoid unnecessary messing about.

One very neat thing about Spur shelving is that you can use little 'shelfettes' anywhere you like, across any number of uprights, to make the maximum use of the space between your main shelves. Halfway between our main work surface and the next shelf up, we have two extra brackets supporting a 22-inch wide shelf, which houses a rackmount Kenton MIDI-CV interface and Emagic Unitor8 MIDI interface, close to our Mac computer. This gets them off the work surface and means that their controls are very easily to hand. Pretty much by accident, our metal uprights ended up being 19.5 inches apart (measured from the inside edges of the uprights), so studio-standard 19-inch rackmount units fit really nicely on shelfettes. Bear this in mind when you space your uprights.

Think ahead about the height of your main work surface. If you're careful about this, you may be able to fix it so that a mobile 19-inch rack on castors can just roll under the work surface when not in use. We already had a mobile rack, which was rather too high to do this, but advance planning may let you make use of the idea. What did work out rather well was our main workstation synth positioning: when this is on its simple X-stand at a nice height ▶

TOM R.WT

20mm

150mm

300mm

8 SCREWS

MDF

Narrow shelves under the main work surface give lots of room for books, software boxes or tapes, whilst not getting in anyone's way.

SHELVING STATISTICS

Pay attention to what the manufacturer of your shelving system recommends when it comes to spacing of uprights and brackets, and shelving load capacities. Leaflets are almost always available where you buy the bits, and it's important that you adhere to the guidelines in them, otherwise your shelves will bow, or the uprights will start to come away from the wall — or both.

▶ for playing, it just pushes under the work surface when not needed, which looks tidy and gets it out of the way completely.

When you begin wiring up your gear, make maximum use of 'snakes' or 'looms': traditionally, you'd buy a drum of multicore cable and solder the connectors yourself, creating a custom wiring loom. But if you don't have the skill or patience to undertake this particular bit of DIY, you can choose from any number of ready-wired snakes. Typically, you'll find them in 8-way or 16-way configurations, with jacks on both ends, jacks on one end and phonos on the other, and so on. They come in varying lengths and they really keep wiring mess to a minimum (SOS sell some nice ones, in the Hosa range; an example price would be £27.90 for 3-metre phono-to-phono, phono-to-jack or jack-to-jack snakes). Other options include MIDI snakes (ideal if you want your MIDI interface near your computer, while keeping sound modules and samplers in a distant rack), insert point snakes and EDAC looms for use with Alesis's ADAT digital multitrack (if you use the EDAC multi-pins, all the ADAT's ins and outs are reduced to a single, highly compact connector). Also, as with any medium-to-large studio setup, seriously consider setting up a patchbay system: it makes life so much easier, and although it's a bit of a pain to do from scratch, you feel pretty smug when all you have to do is plug in a couple of little patchcords to connect a noise gate to the insert point on mixer input 13, instead of footling around at the back of the mixer and rack all the time. Once again, a little planning will go a long way: for example, if this is the first time you've wired up a patchbay, start by counting all the connections you'll be bringing out (taking note of where non-normalised or semi-normalised operation would be appropriate) in order to ascertain the size and number of patchbays required. Even draw a wiring diagram if it'll help you conceptualise the end result. Paul White's recent article on wiring patchbays, in SOS March 1998, will prove invaluable during your planning stages.

The whole system took us less than a weekend to put together and re-wire, and is showing no sign of strain at all, despite having been in use for many months. It cost just under £300, but that total also includes four extra uprights and the brackets and shelves needed to fill an alcove elsewhere in the room which is now used for storing magazine binders and more books. This isn't dirt-cheap, granted, but when you bear in mind that a dedicated studio table, from one of the commercial manufacturers, could set you back almost that much by itself, the system starts to look pretty cost-effective. And anyway, aren't you worth it? **SOS**

MANUFACTURERS & SUPPLIERS

• Studiospares can sell you practically everything you need for your studio, including leads and plugs, patchbays, and a selection of racks and rack accessories. Write or call if you'd like a copy of their catalogue.

A 61-63 Rochester Place, Camden Town, London NW1 9JU, UK.
T +44 (0)171 482 1692.
F +44 (0)171 485 4168.

• Patchbays: the following companies make patchbays that have been reviewed in past issues of SOS.

A Neutrik UK Ltd, Columbia Business Park, Sherbourne Avenue, Ryde, Isle Of Wight PO33 3QD, UK.
T +44 (0)1983 811441.
F +44 (0)1983 811439.

A Prosonix, PO Box 32, Hailsham, East Sussex BN27 3XT, UK.
T +44 (0)1323 848808.
F +44 (0)1323 441185.

A Isotrack, PO Box 747, Poole, Dorset BH12 4YG, UK.
T +44 (0)1202 747191.
F +44 (0)1202 747191.

A P&R Audio, Technology House, Stone Cross Farm, Lewes Road, Laughton, East Sussex BN8 6BN, UK.
T +44 (0)1323 849522.
F +44 (0)1323 849533.

A Kelsey Acoustics Ltd, 27 Beethoven Street, London W10 4LL, UK. (Kelsey sell lots of other studio supplies; call them for a catalogue.)
T +44 (0)181 964 8000.
F +44 (0)181 964 1010.

• Mobile racks: Studiospares sell a couple of mobile racks, but you can also check out the following:

A Brandoni Music Ltd (for the Quicklok range of stands, rack equipment and studio furniture),

Unit 3-6, Wembley Commercial Centre, East Lane, Wembley Middlesex HA9 7XJ, UK.
T +44 (0)181 908 2323.

• All sorts of electronic components and accessories, including the miniature vaccuum cleaner, can be obtained from Maplin Electronics. You can buy their catalogue in WH Smith, and it's available on a Windows/Mac compatible CD-ROM, for just £1.95, as well as in paper form (be warned, though, that we found the CD-ROM rather inaccessible and unsatisfactory when compared to the traditional catalogue).

A Maplin Electronics plc, Maplin House, 274-288 London Road, Hadleigh, Benfleet, Essex SS7 2DE, UK.
T +44 (0)1702 554002.
E techsales@maplin.co.uk
W www.maplin.co.uk

FOSTEX

CD quality digital audio. Better than digital mixer. Media independent. With knobs on!

Recorder Section
- True non compressed audio ensures outstanding quality
- Media independent: Can record to external SCSI device or to internal E-IDE hard drive (optional)
- Extra L & R tracks. Stereo mix-bounce of 4 main tracks
- Track Exchange. 4 main tracks and L&R tracks can be exchanged
- Copy Paste, Move Paste & Erase. Non-destructive editing with Undo/Redo
- ±6% vari-pitch
- Digital input from 2-track source (CD, DAT, etc.)
- 2-track Digital out
- Digital output of up to 12 sources in remix
- MTC Slave, MIDI Clock & SPP out & MMC

Mixer Section
- Real knobs - no multiple button pushes to make simple adjustments
- 4 main input plus 4 monitor channels, 2 AUX sends & 2 stereo AUX returns
- Balanced XLR mic inputs (Ch 3-4) with trim control
- 3-band EQ with wide-band parametric MID

*Requires optional E-IDE fitting kit

Some things never change...

If you're reading this magazine, chances are you're a musician. - Not a rocket scientist, not computer wiz, and certainly not a geek.

So why is it that other manufacturers expect you to have a degree in button pushing just to get at the mid-band EQ or to adjust the aux send level of track 2?

With the new Fostex FD-4 you'll have no such hassle. It features a mixer with >105dB dynamic range (which is better than digital), 3-band EQ, balanced XLR mic inputs, insert points and 2 aux sends.

A real mixer. With knobs on!

And if you appreciate that, you'll flip when you hear that the FD-4 offers all the other tools you need to make fantastic and professional, uncompressed digital multitrack recordings while giving you the freedom to choose which media to use. Either an external SCSI device such as the SyQuest or ZIP drive you bought for your sampler/computer, or even internal E-IDE hard drive*.

4 tracks of genuine CD quality (16-bit linear @ 44.1kHz) recording, easy copy, move & erase editing with undo (just like a word processor), built-in 2 track mastering mode, flexible sync facilities and heaps more.

Fostex have long had the knack of stuffing their multitrackers with innovative features, yet making them incredibly easy to use.

They also insist that your recordings are not compromised by compression techniques.

...some things never change.

unpack connect scsi device format plug in

FD-4 quick start guide

Stereo Out L/R
Monitor Out L/R
AUX Send 1/2
AUX Return 2
AUX Return 1
Insert Ch 3 & 4
XLR Mic In Ch 3 & 4

DATA I/O
Punch In/Out
MIDI In/Out
Recorder In L/R

SCSI
Interface

MEDIA INDEPENDENT - STANDARD SCSI-2
You choose which type of media you want to use from removable systems such as SyQuest EZFlyer, Zip & Magneto Optical to large capacity hard drives

FD-4
DIGITAL MULTITRACKER
FOSTEX

FD-4
Digital 4-track Recorder

Exclusively distributed in the UK by SCV London
6-24 Southgate Road London N1 3JJ
Tel: 0171 923 1892. Fax: 0171 241 3644
email: fostex@scvlondon.co.uk
web: http://www.scvlondon.co.uk

adjust levels

record

mix

PAUL WHITE finds out whether Tannoy's new Reveals have anything to hide...

Tannoy are most often associated with dual-concentric speakers, where the tweeter is placed at the centre of the bass/mid driver, with a view to producing an accurate point sound source. The concept has considerable merit but invariably forces compromise in other areas of the engineering design, the outcome of which is that dual-concentric speakers tend to have a characteristic sound rather than being ruthlessly neutral. Good dual-concentric speakers are also costly to build, so when they're designing more affordable monitors Tannoy occasionally revert to the traditional separate tweeter and bass/mid driver topography, often with considerable success.
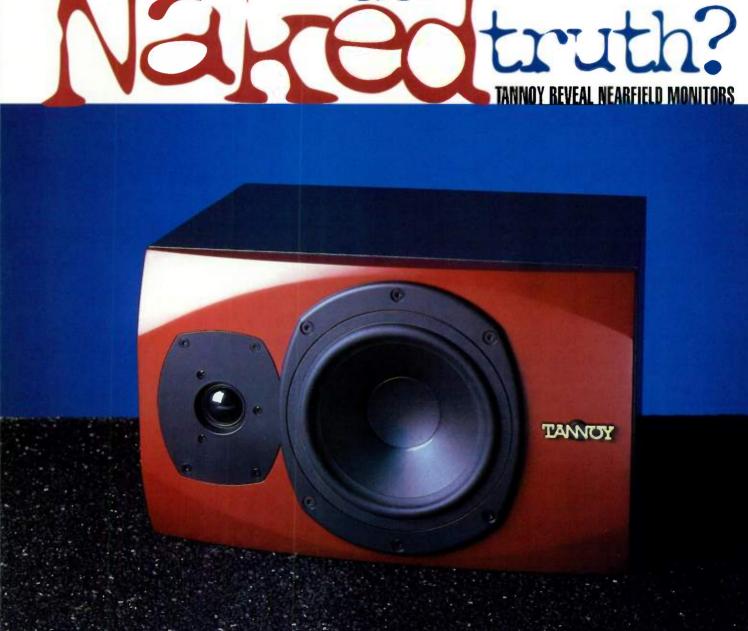
SHOW & TELL

The Reveal monitors we're looking at here fit in with this alternative approach for Tannoy — they're passive, 2-way, nearfield monitors using discrete drivers in a rear ported cabinet. Up to 3kHz the bass and mid-range is handled by a 6.5-inch, synthetic-coned driver in a roll surround, which hands over to a 1-inch soft-dome tweeter after this point, though no details regarding the crossover slope or filter characteristics are provided. Both drivers are magnetically shielded, so they can be used near computer or TV monitors. Because their overall sensitivity is a reasonably high 90dB for 1 Watt at 1 metre, in a typical half-space situation (solid wall behind the speakers), amplifiers of between 50 and 100W per channel provide adequate power. Overall, the Reveals measure 340 x 210 x 260 (mm) and weight 7kg each. The MDF cabinets are finished in a grey suedette vinyl foil.

TAKING OFF THE WRAPS

Though the Reveals are largely conventional in concept, they feature several design points worthy of note, not least the convex dust cap at the centre of the bass/mid driver. Opinion is divided between designers on whether this is a good or a bad thing, but Tannoy obviously put it there for a reason. The distinctively profiled front panel is machined from 40mm MDF, making the whole assembly very rigid, and the port is located at the rear of the cabinet,

the Naked truth?

TANNOY REVEAL NEARFIELD MONITORS

both to conserve panel space and to reduce port noise. The baffle profiling will help reduce cabinet-edge diffraction, but its main function is probably styling — hence the fetching shade of maroon! The flush-mounted, soft-dome tweeter (Tannoy model 1205) is positioned as close as possible to the bass/mid unit (Tannoy model 1690), in order to maximise the vertical angle over which the speakers produce a coherent image. Conversely, the bass/mid driver appears to be surface mounted, but the chassis surround has a rounded edge to avoid abrupt transitions that might cause diffraction. Rear connection is via colour-coded binding posts that can also accept banana plugs, so there's no need to struggle with Speakon connectors. The crossover is fixed to the back of the terminal plate.

I'm pleased to say that the manual that comes with the Reveals is both informative and amusing. It covers all the basics of speaker placement in a studio environment, including warnings against placing pot plants in front of them, and also offers practical advice on simple ways to improve your

> ## "Overall, the speakers are tonally well balanced, and while the bass end lacks the extension of bigger monitors, it is still tight and quite solid."

monitoring environment, using little more than string and sticky-backed foam — really! It's a little short on technical information, however, so although you get the basic specification, there's little detail in areas such as crossover design and driver construction. For example, we're told that the monitors' nominal impedance is around 6Ω and that the frequency response extends from 60Hz to 20kHz, but as we're not told how many dBs down the response is at these points, the figure is absolutely meaningless. Neither is the maximum SPL specified. From the frequency graph supplied, the response appears to be around 3dB down at 90Hz, 10dB down at 60Hz and barely down at all at 20kHz. Judging by the impedance plot, the cabinet is tuned at around 90Hz.

REVEALING ALL

Given their relatively low cost, the Reveals turned in a very respectable performance in listening tests. My test material showed just a slight tendency towards hardness in the upper mid-range, when compared with the ATC SCM20s that I use as a reference, but the soft-dome tweeters help keep the high end adequately smooth so that you can work for reasonable periods without experiencing fatigue. The sound has a few very minor rough edges to it,

but the general impression is one of clarity and detail. Overall, the speakers are tonally well balanced, and while the bass end lacks the extension of bigger monitors, it is still tight and quite solid. The stereo imaging is also good, even some way off-axis. Of course, you have to use these speakers in 'portrait' rather than 'landscape' mode to maintain good imaging — unlike Tannoy's dual concentrics, which can be used any way up. The manual explains this quite thoroughly, presumably to try to sell you the idea of buying dual-concentric Tannoys next time around, but I bet we'll still see Reveals perched on their sides on more than one meter bridge!

Discerning listening tests on vocals revealed only slight traces of edginess at the upper reaches of the voice, and while this trait can increase the apparent amount of sibilance in some voices, it certainly isn't a major concern for a speaker of this price — and, indeed, this slight edginess is a lot less pronounced than on some monitors costing many times the price. The massive baffle pays off in keeping the sound tight and well focused, and although there is a certain lack of finesse when you do a direct comparison with a really good reference speaker, the ears soon compensate. What's important is that, in terms of 'broad strokes', the Reveals present a well balanced and largely accurate impression of how a recording sounds, and they seem fairly forgiving of imperfect listening rooms If you can't afford to budget more on monitoring, the Reveals have to be one of the better value small monitors around. **SOS**

three-stage process. We gather PCM [sounds] from everywhere, and we know essentially what we're looking for — in addition to the meat and potatoes stuff, that is. We look for interesting things, like pot tops or whatever, then we get together and look at the material we've got. In the case of the Trinity, we had two string files where one was more classical and the other was more for pop applications, and we couldn't decide between the two of them. So we decided that we'd reduce their size somehow and use both of them. At that stage, the sounds are burned into ROM for the prototypes. Then we go back to our respective countries and make programs for guitars, pads, electric pianos or whatever, using these waveforms.

"The next stage is that we meet up again in Japan, and one day we'll do the guitar patches, another day we'll do another sound type. For example, we need a guitar, a fretless bass, and a slap bass — somebody might have something slightly unusual that's interesting, then we fight and go through the politics until we arrive at the set of sounds that will actually go into the instrument. Once we've decided on the programs, you then go back and make combinations of the patches — piano and strings, and so on — then we meet again to decide which ones to use."

Where do the more abstract sounds, such as pads that evolve or tinkle or do other strange things, come from? Do you have any examples of how these things are put together?

SM: "I've never been a programmer in the sense of saying, 'I'll take this sine wave and make a piano out of it'. I've always started out with the PCM waveforms and then tried to use them in interesting ways. Years ago, on the T-series, I made a patch

Making

Waves

Do you ever wonder who's responsible for the waveforms and sound patches in your favourite synth? If it's a Korg keyboard, then the answer might be Steve McNally, one of their team of sound developers. PAUL WHITE talks to him about his work.

Steve McNally is part of the small in-house team at Korg who have been responsible for filling the company's keyboards with factory sounds since the M1 days, and his most recent work can be heard on the i30 keyboard. The team of around five or six sound designers includes people from around the world — an international mix makes perfect sense if an instrument is to have worldwide appeal.

Steve first worked for Korg in Canada, but is now employed as a consultant to the company in Japan, designing both sounds and accompaniment styles for a range of keyboards; and when he's not designing sounds, he's travelling the world showing them off at trade shows and demonstrations. I was curious to know at what level sound designers get involved in a product — are they confronted by a finished machine with empty patches, or do they also have a say in which PCM waveforms are built into the machine in the first place?

Steve McNally: "If we were to look at the way it worked for the Trinity, for example, it's a kind of

called 'Dusty Sax'. It was a combination program, but there was this fluttery pitch-enveloped noise which I layered with the sax. Next, I pitched the fluttery sound up by a fourth, but it was a bit in the way, so I thought I'd try routing it into the reverb and then using it completely wet. That way I ended up with a sax that had this tinkly reverb about a fourth up, and it was like 'wow!'. But you never really set out with that kind of end result in mind, they just happen, and weird things are kind of my speciality. If the other guys want something weird, it's a case of, 'let's bring in Steve's stuff!'."

What kind of PCM sounds do you have to choose from, apart from the obvious instrument sounds and voices? Do you get bamboo chimes and people scraping piano strings, or what?

SM: "Sure, and we did some interesting PPG-type things to some of them. I don't know if you remember that thing called 'Flutter' — I think it was on the X3, and it was a loop of something one of the guys had come up with from joining things together. Again in the T-series, there's a particular sample of

a pot lid, and it was just gorgeous. You could play that as a tuned instrument and it was stunning — almost an electric piano type of sound."

Pretty much every synth you buy now uses PCM samples as the raw sound elements, but in the past Korg have produced ground-breaking instruments that worked on different principles, for example the Wavestation, and the application of physical modelling in the Prophecy and Z1. Is physical modelling likely to re-emerge in a low end machine?

SM: "If you look at what the Prophecy cost for a monophonic synth, you can now have a 12-voice expandable Z1, which is far more powerful, for not that much more money. The difficulty is knowing where to place these instruments. Take the Wavestation: I was in love with that thing, and I'd sit down with people and show them how to put together really interesting wave-sequenced sounds, add vector control and velocity switching, and they'd go 'let me hear the piano'. I think that there was a period where in pop music when there were lots of great keyboard sounds around that people just weren't using. The great thing about dance is that people are more experimental with their sounds, and that's now filtering over into pop music."

But surely, even when interesting keyboard sounds weren't being used on pop records, you could still hear them everywhere on film and TV commercials. Wouldn't it be practical to take some of the more experimental elements from these older instruments and combine them with the more conventional sounds that people expect to find? Without something that's new, everything converges on the GM sound set.

SM: "When we were developing our physical modelling technology, we had this thing that could deal with virtually any type of synthesis, and the first thing we did was ask if it could do wave-sequencing, but wave-sequencing is pretty intensive in the amount of processing power it needs. However, I agree totally — I loved that ability to shape sounds. What happened with the Wavestation is rather interesting in that it was really just after the M1, and we wanted to make a synth that wasn't PCM playback — a kind of anti-M1. It wasn't until later that Korg expanded the Wavestation to have pianos, guitars, drums, etc. I think that made the instrument more interesting."

Do you think that samplers have taken over the role of producing abstract sounds, or do you think there's still a need for that capability in synthesizers?

SM: "Ever since the M1 there's been the problem of deciding what to make next, and even though we might not have any ideas at the time, we seem to come up with something new to put into the next model. Guys that have samplers don't seem to be that interested

in this type of instrument anyway. If they're going to do a piano track, they'll load up a 64Mb piano sample and still have room for huge, unlooped drum sounds."

I've often felt that, since the M1, PCM-based instruments haven't advanced much, other than to sound cleaner and perhaps offer more sounds.

SM: "In some ways that's true, but the Trinity is a different story. With the flexibility of the effects, sound quality and the Physical Modelling option you get a real improvement from the typical PCM keyboard."

Do you see any advantage in adopting more complex filter types, rather than the usual resonant low-pass synth filters, to help articulate sounds?

SM: "Yes, although putting on my marketing hat, just saying that something has moving filters isn't going to make people want to go out and buy it. The keyboard has to have more than that."

But surely these days you expect a good GM set in any instrument, almost regardless of what it is, so why not give them that plus the 'added interest' features in the same machine?

SM: "I think that's a good idea. Perhaps the Wavestation would have been even more successful if we'd had drums, piano, guitar and bass in it right from the beginning."

So what sells a keyboard now? Is it just a good set of GM sounds, or is there more to it? After all, there must be countless instruments that do a good job of delivering the standard sounds.

SM: "It's true, though I've always felt that in the past Korg have produced better sounds than the competition. These days you can walk up to any keyboard and it will sound great — and even if you don't like the piano you may think the guitar is better, so it's not so easy to say which [keyboard] is best. Still, comparing the Trinity with anything else in its price range, I think it sounds better than the other stuff."

FROM TUBAS TO TECHNO

When you're selling the same basic instrument around the world, how do you reconcile the different musical cultures and requirements of different countries? The Germans may want tubas and accordions where the Americans want country music sounds and we in Europe want techno.

SM: "It is very difficult, but that's one of the reasons the voicing team is international. If the voicing team was American, somebody might ask why there's only one accordion, and the answer would be, 'because we had to put it in there'. But a German sound designer might want to use five or six of them. We can put enough stuff inside a keyboard now to cover all these areas — it's really a matter of how you get to it."

Is the finite amount of sample ROM available a big problem? And if so, why don't instruments have flash ROM so that

Korg's Z1: "At the development meeting it was initially described as a polyphonic Prophecy."

"...just saying that something has moving filters isn't going to make people want to go out and buy it."

▶ a different set of sounds can be loaded into the same machine depending on what country it's being sold in. In fact, why not produce a CD-ROM of PCM sounds so that the end user can, in effect, choose their own PCM sound set? That way the Germans could have their 10 tubas and 20 accordions.

SM: "Occasionally, different countries load in their own programs for their market, like with the Z1 here in the UK. We've talked about this, and the concept is very attractive, but the reason you get so much from a modern machine is that the chips tend to do several jobs at the same time. If you change that so you can put in anything you want, you have to walk the fine line between the instrument being a synth and a sampler."

But surely this would combine the flexibility of a sampler with the instant patch access of a synthesizer?

SM: "It's a big job collecting that many good PCM sounds to put on a disk. It's not only a matter of collecting the sounds in the first place — you also have to loop and process them to make the samples as small as possible, so you can get a lot of different sounds into the ROM that still sound good. Then you need to make programs and combinations. It's a very big job."

MODELLING:
A LEARNING EXPERIENCE

Returning to the subject of modelling, I've often said that I feel modelling works better on monophonic sounds than on polyphonic sounds. Where do you see modelling helping on polyphonic instruments?

SM: "The Z1 is pretty much what we're talking about here, and at the development meeting it was initially described as a polyphonic Prophecy. But when you actually sit down and play a Z1, although it can do all the great monophonic Prophecy stuff, that's really just the beginning, and the polyphonic sounds are very different. When you've played the electric piano on a Z1, then go back and play it on a basic PCM machine, it's apples and oranges. The way the timbre changes as you alter your playing dynamics is far more realistic. And the synth stuff is great. Once you can do FM polyphonically, it sounds so much cooler. It's like when synths became polyphonic in the old days, their capabilities soared. I think it's the same
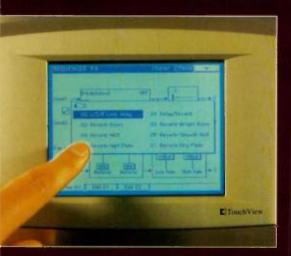
The Korg Prophecy monosynth offers a number of different synthesis methods, including physical modelling.

with the Z1. But you stand behind these guys at trade fairs and they stab away at a single note, and run through the patches as though they were checking out an M1. That's not how you check out the controllability of modelling. It's an educational thing where we have to encourage musicians to play these instruments in a different way, and that's the scary thing. When you sit down and play a PCM synth, everybody knows the rules, but Program B25 on the Prophecy might use a certain wheel to control the breath content of a sound, while on Program 17, the same wheel might add growl. There's no consistent user interface, so each patch is a learning experience in its own right. But that's part of the fun of it."

What would you like Korg to build next, if you had your own way, and how do you think the user interface can be improved now you're using touch screens?

SM: "I don't really know what we'll make next. As I said, we all sit down at these meetings, and wonder where we should go. I would really like to see something, along the lines of what you're suggesting, that covers the meat and potatoes thing but also has new and more interesting capabilities for changing sounds.
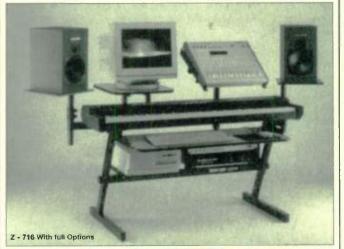
"I like the touch screens. I've used everything, but now I've got used to touch screens. I start poking away at an X3 screen waiting for something to happen! Some people ask for monitor outputs and mouse ports, and I guess there is that, but for me, touching a screen is faster than working with a mouse." **SOS**
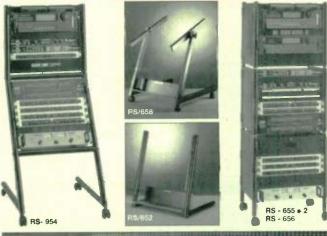
"I like the touch screens... for me, touching a screen is faster than working with a mouse."

Arboretum's latest Hyperprism suite of effects and processors offers no less than 29 modules, at an average price of less than £10 each. JANET HARNIMAN COOK gets plugged in.

The Arboretum *Hyperprism* suite of professional sound design tools has been a favourite of audio engineers in the film, broadcast, games and music recording industries since 1995, when *Hyperprism* made its debut as a plug-in for the Mac-based Digidesign ProTools TDM system. *Hyperprism* was used in the soundtrack recording of motion pictures including *The Fifth Element*, *Contact*, *Pulp Fiction*, *Speed*, *Apollo 13* and *The Flintstones*, and features on recent album projects by recording artists such as Nine Inch Nails, Duran Duran, White Zombie and Public Enemy.

Hyperprism is now available for Windows 95 Pentium PCs as well as Macs, and is the latest megastar to join the DirectX audio plug-in firmament which, over the last few months, has witnessed the appearance of stunning studio-quality software from major developers including Waves, Steinberg, Sonic Foundry, Opcode, TC Works and Power Technology.

SUITE 29

The *Hyperprism Plug-in Pack* comprises 29(!) audio processing modules. These not only cover most essential studio processing requirements, but are also capable of exotic routines more typically encountered in the realms of sound design and synthesis. Audio data is processed with 32-bit floating-point internal precision, to ensure pristine sound quality.

The PC version — *Hyperprism DX* — was used for this review. It conforms to the MS DirectX audio plug-in standard and features a real-time preview function so that parameter changes can be auditioned as you make them. Like all DirectX audio plug-ins, *Hyperprism DX* requires a compatible host program, and *Cubase VST* 3.553, *Cakewalk Pro Audio 7*, *Sound Forge* 4.0d, *WaveLab* 2.0 and *Cool Edit Pro* 1.1 were used to run *Hyperprism* for this review.

THE PACKAGE

Inside the *Hyperprism* box are the software license agreement, the user registration card, and a dual-platform CD-ROM that contains the *Hyperprism* installation files, a collection of audio files for tutorial use, utilities and demos for the Mac, and the on-line manual. The last is in HTML format, and can be ▶

ARBORETUM *HYPERPRISM* PC & MAC EFFECTS PLUG-IN PACK

Hyperprism DX's Hall reverb running as a channel insert in Steinberg's Cubase VST.

The Cakewalk Pro Audio plug-ins menu showing all 29 Hyperprism DX processors.

use of space, as the *Hyperprism DX* interface occupies about 40% of the total screen area on my PC.

The appearance of the *Hyperprism DX* modules varies slightly depending on the host application that is used; the interface consists of the distinctive Blue Window gesture-orientated control panel, the output level faders (+/-18dB), the Preview, Process and Preset buttons, and the horizontal parameter faders, which feature user-definable range scaling, to enable more accurate targeting of parameter hot-spots than can be achieved using the default range settings. I was slightly surprised by the absence of level metering, but this is not a huge problem, as you can use the meters provided by your host application.

THE BLUE WINDOW
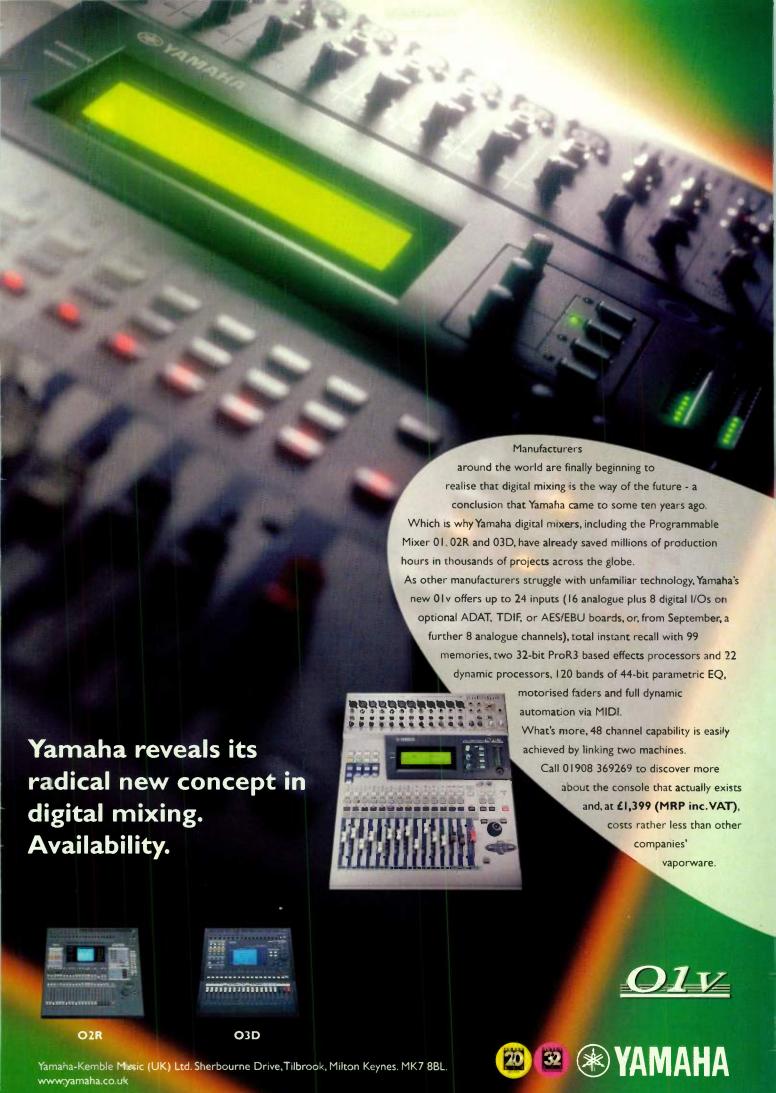
Depending on the module being used, the *Hyperprism* Blue Window may feature up to six different processing parameters that will react in unison whenever the Blue Window icon is dragged around the blue play zone. Arboretum describe this as their gestural interface environment, and each parameter under Blue Window control may be assigned to respond to either horizontal or vertical pointer movement, with parameter changes reflected by the automatic repositioning of the associated parameter faders. I found the Blue Window control surface efficient and fun to use — it provides a quick way of exploring the sonic capabilities of each processor and, if used experimentally, yields a wealth of creative opportunities for real-time audio manipulation. It is in the area of real-time manipulation that the Blue Window environment shows its true potential. Although Macintosh versions of *Hyperprism* allow you to record Blue Window movements as a sequence, this will sadly not be possible on the PC until a native version of *Hyperprism DX* for *Cubase VST PC* becomes available. Of course, this does not prevent you from recording the Blue Window changes as you make them in Preview mode, either to a new audio file on the PC if you have multi-channel audio capability, or to an external recorder such as DAT or ADAT.

FILTERS

Hyperprism DX offers four types of filter: High-pass, Low-pass, Band-pass and Band-reject. A high-pass filter has a definable cut-off frequency below which frequencies are attenuated, while those above this point pass through the filter unaffected; the steepness of the filter slope is defined by the level of attenuation or boost per octave that is applied at the cut-off frequency: a 6dB/octave slope will result in a gentle roll-off, whereas a 90dB/octave slope results in a steep 'brick wall' effect. A common application for high-pass filtering would be the removal of unwanted ▶

▶ viewed using a web browser such as *Navigator* or *Internet Explorer*. Technical support is available free of charge to all registered owners (email your query to tech@arboretum.com), and telephone support is available directly from Arboretum Systems or through their UK distributor Unity Audio. If you have access to the Internet you can download, free of charge, fully-working versions of the *Hyperprism* Ring Modulator, Echo and Vibrato plug-ins from the Arboretum Systems website.

WHAT'S UP DOC?

I'm not a fan of on-line manuals, and when faced with learning large applications I would much prefer a printed version. The on-line *Hyperprism* manual is mercifully short, covering the installation routine and the details of the various parameter functions that control each *Hyperprism* processor, and providing a few simple tutorials using wave files included on the CD-ROM. Experienced users should experience little difficulty in learning the various *Hyperprism DX* routines, and even newcomers should fare well, thanks to extensive Windows Help and the fact that most modules contain a selection of factory presets.

HYPERACTIVE!

Installation went without a hitch: the *Hyperprism* modules duly appeared in the DirectX menus of the different host applications, and no compatibility problems were encountered. Windows 95 users may find the *Hyperprism DX* workspace a tad stark and utilitarian — no psychedelic plasma level meters or sexy 3D graphic modelling here! — but the interface is uncluttered and easy to negotiate, although I would prefer a layout that made better

Yamaha reveals its radical new concept in digital mixing. Availability.

Manufacturers around the world are finally beginning to realise that digital mixing is the way of the future - a conclusion that Yamaha came to some ten years ago. Which is why Yamaha digital mixers, including the Programmable Mixer 01, 02R and 03D, have already saved millions of production hours in thousands of projects across the globe.

As other manufacturers struggle with unfamiliar technology, Yamaha's new 01v offers up to 24 inputs (16 analogue plus 8 digital I/Os on optional ADAT, TDIF, or AES/EBU boards, or, from September, a further 8 analogue channels), total instant recall with 99 memories, two 32-bit ProR3 based effects processors and 22 dynamic processors, 120 bands of 44-bit parametric EQ, motorised faders and full dynamic automation via MIDI.

What's more, 48 channel capability is easily achieved by linking two machines.

Call 01908 369269 to discover more about the console that actually exists and, at £1,399 (MRP inc. VAT), costs rather less than other companies' vaporware.

O2R

O3D

01v

YAMAHA

KORG

Heard the great sound of Trinity?

We think you have.

Trinity, The most played Professional Workstation, anywhere.

TRINITY
MUSIC WORKSTATION DRS
S E R I E S

ARBORETUM *HYPERPRISM*

Probably the most natural sounding DirectX reverb in the world — the Hyperprism Hyperverb.
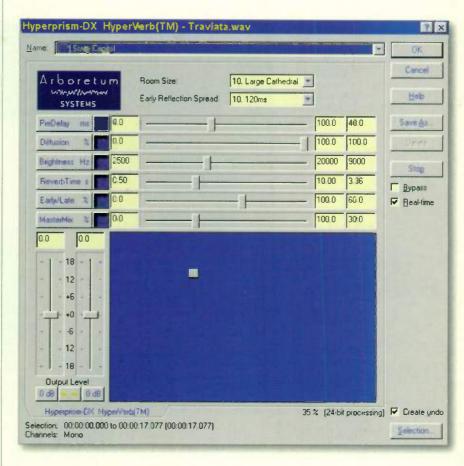
► low-frequency components such as bass rumble. Low-pass filters work in the same way, but spectral elements above the defined cut-off frequency are rejected while those below remain unchanged, so low-pass filtering may be used to remove hiss and other spurious high-frequency sounds. Rather than having a single cut-off point, like high-pass and low-pass filters, a band-pass filter has a definable frequency range that can be boosted or attenuated, and signals outside of this bandwidth pass through the filter unchanged; a band-reject filter acts in the opposite way — the defined bandwidth is excluded from the filtering process. The *Hyperprism* filters feature variable Q and smoothing, and are useful for a variety of basic studio equalisation functions but, sadly, more complex multi-band processing is at present beyond the scope of *Hyperprism*.

DELAY AND REVERB

Three delay processors — Single, Multi and Echo — are available in *Hyperprism*. The first is a simple single repeat with a maximum two-second delay time; Multi provides a three-element delay with user-defined times of up to two seconds for the first two repeats and the third delay time determined by the sum of the two; Echo is identical to Single Delay, but with the addition of a feedback parameter. This produces a richer, fatter multiple-delay texture by mixing a proportion of the output signal with the incoming audio.

The *Hyperprism* manual makes great claims for the Hyperverb — and with justification, for Hyperverb ►

PC REQUIREMENTS

Hyperprism DX is a Windows 95 native application and is not compatible with earlier versions of the MS Windows operating system. To run it you will need a DirectX compatible host application and a Pentium PC equipped with an Intel 166 MMX (or better) processor, at least 32Mb RAM and a fast EIDE, UDMA or SCSI hard drive.

The reference PC used for this review is an Intel Pentium 233MMX with SuperMicro ATX motherboard, 512k pipeline burst cache, 64Mb SDRAM, 4Mb Virge DX PCI graphics card, running 1080 x 868 x 64k colours on a 17-inch monitor, with Creamware TripleBoard & Turtle Beach Fiji audio cards, Adaptec 2940 PCI SCSI card, Fujitsu EIDE & UDMA AV hard drives, and a Logitec Marble optical trackball (very cool for Blue Window control).
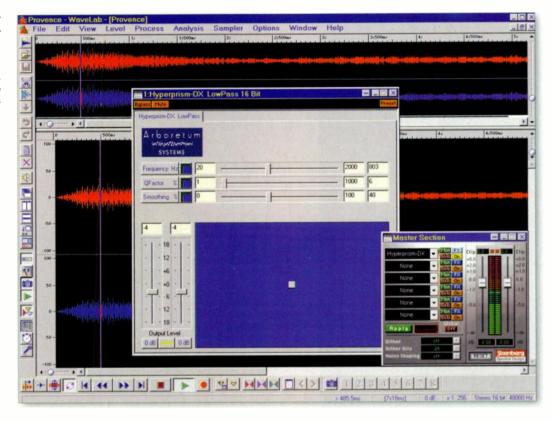
Hyperprism DX offers four equalisation processors. Shown here is the Low-pass Filter running in Steinberg's WaveLab.

"I found the Blue Window control surface efficient and fun to use — it provides a quick way of exploring the sonic capabilities of each processor."

is one of the clearest, most natural sounding reverbs I've ever heard, and although both the simpler Hall and Medium Room reverb algorithms sound very good, the acoustic ambience created by Hyperverb really is quite remarkable in its realism.

Six Hyperverb parameters (pre-delay, diffusion, brightness, early/late reflections balance, reverb time (decay) and master wet/dry mix) can be assigned to Blue Window control, and in addition the room size and early reflection characteristics can be adjusted.

MODULATION PROCESSORS

Hyperprism DX features eight modulation processors, as follows:

- **Phaser:** This produces a good range of effects, including a sweet, whispering phase shift that I found similar in character to that produced by vintage guitar devices. It features variable delay (0-2000mS), depth, feedback and base frequency (which defines the frequency at which processing starts).
- **Flanger:** The Flanger works by adding an LFO-controlled variable delay to the input signal and mixing this with the incoming audio to produce a comb-filter effect; you can control the speed, depth, feedback and mix to produce a wide range of sounds, from gentle, animated filter sweeps to more intrusive, resonant effects. In my opinion, this is one of the best plug-in flangers on the market, and it reminded me of the lush comb filtering I used to get from my vintage Electro Harmonix Electric Mistress foot pedal — albeit without the noise and distortion!

- **Chorus:** The *Hyperprism* Chorus is capable of producing lush textures when used on voices and with pad sounds, but the lack of presets makes it difficult to assess its full capabilities.
- **Tremolo & Vibrato:** These effects are created by applying, respectively, amplitude or pitch modulation to the input signal. Both processors perform well, and may be used, for example, to add expression to keyboard lead lines and string pads.
- **Ring Modulator:** This is an oddball effect that is typically enharmonic and metallic in texture. The input signal acts as a carrier, which is modulated by a second signal to generate a composite signal made up from the sum and difference of the carrier signal and the modulator. If the carrier signal is 200Hz and the modulator is 50Hz, for example, the composite output signal will be a mix of signals at 250Hz and 150Hz. The Ring Modulator is great for creating special effects such as robot voices, and can produce spectacular results when used to process drum and percussion tracks.
- **Frequency Shifter:** Similar effects to the Ring Modulator can be obtained with the Frequency Shifter, which maintains the harmonic series as it moves along the spectrum. The lowest frequency in the original sound still corresponds to the lowest frequency in the new sound and no new low sidebands are created, as with the Ring Modulator.
- **Vocoder:** The *Hyperprism* Vocoder is a little unusual in the way it works: to prepare material for processing, you must first create a stereo audio file with the carrier signal in one channel and the modulator signal in the other. The Vocoder analyses ▶

The Limiter at work in Sound Forge.

quality as the rest of the suite and although not possessing the same versatility and power as, for instance, the Waves *C1+*, they nonetheless perform the simpler studio dynamic processing tasks well enough.

Two modules complete the set. The Pitch Changer adjusts the frequency of the input by plus or minus 200% (+1 octave) and, like many of the *Hyperprism* processors, provides tremendous scope for effects designers. The same goes for the Sonic Decimator, which degrades the input signal by simulating the effect of reducing the bit-rate of the audio sample, complete with a reduction in bandwidth and dynamic range, and an increase in noise and digital distortion. Tomorrow's Retro today, anyone?

CONCLUSION

The new modules available in v1.5 of *Hyperprism DX* (which also corrects a couple of bugs from v1.0) are a valuable bonus — especially the Hyperverb and the Vocoder — and the new factory presets and Windows Help are also very welcome. However, this pack faces fierce competition in the PC audio plug-in market, not only from processing bundles such as Waves *Native Power Pack*, Power Technology *DSP Effects* and Sonic Foundry *XFX 1* and *XFX 2*, but also from the high standard of onboard processing available in host applications such as *Cubase VST*, *WaveLab* and *Sound Forge*. Time-stretching, multi-band equalisation and frequency-conscious dynamic processing are conspicuous by their absence, but despite this, *Hyperprism DX* 1.5 is unarguably the most versatile DirectX bundle currently available, and makes the ideal complement to the processing facilities provided by the Waves *Native Power Pack*.

To sum up, *Hyperprism DX* is a true professional quality plug-in suite that can be safely recommended. It should soon become an industry standard in PC-based audio production facilities involved in mastering, broadcast and film post production, multimedia, game design and music recording.

► the modulator signal and applies imitative 26-band filtering to the carrier signal; adjusting the modulator parameter allow the carrier and modulator roles of each channel to be reversed. This processor is very musical and is capable of producing pristine-quality classic vocoder sounds.

STEREO MANIPULATION

Six methods of stereo manipulation are featured in *Hyperprism*:

- **Pan & Auto Pan** shift the sound across the stereo field, Pan simply placing the sound in a fixed position while Auto Pan employs an LFO envelope to dynamically shift the stereo position of the sound during playback.
- **Quasi Stereo** imparts stereo qualities to a mono signal using complementary comb-filtering techniques.
- **More Stereo** extends the perceived boundaries of the stereo image beyond the left and right speakers, creating an enhanced sense of space.
- **M+S Matrix** provides virtual M-S decoding facilities, and Blue Window controllers can be used to tweak the apparent width of the stereo image.
- **Stereo Dynamics** claims to dynamically locate the input signal front to back in the stereo field, as well as left to right, but I confess to being unconvinced — perhaps the process is too subtle for my ears!

DYNAMICS

Hyperprism includes easy to use Noise gate and Compressor modules, plus a rather good Limiter. These effects exhibit the same high processing

"Hyperverb is one of the clearest, most natural sounding reverbs I've ever heard."

Room for Improvement

PRACTICAL ACOUSTIC TREATMENT

PART 2: In his second article on acoustic treatment, PAUL WHITE tackles the absorbing subject of acoustic traps.

L ast month's introduction to acoustic treatment pointed out the importance of room dimensions in achieving an even modal response, but that doesn't help much if your room is already built. Fortunately, by taking a pragmatic approach, combined with the use of nearfield monitors, you can produce well-balanced pop music in a relatively unsophisticated room. When recording, there's usually a lot of close miking employed, and even then much of the sound subscribes more to fashion rather than to pure fidelity. It's really when recording speech, vocals or acoustic instruments that well-behaved acoustics are most important in the actual studio area. Equally, an apparently problematic control room can usually be set on the road to workability by choosing appropriate monitors and mounting them in a suitable position. Further improvements can be made by placing suitable acoustically absorbent materials within the room to absorb sound that would otherwise be reflected by the room boundaries, so that you hear more of the direct sound from your monitors (reflected sound

is misleading, as it will be coloured — effectively, EQ'd — by whatever it has bounced off before it reaches your ears).

ABSORBING STUDIES

Having introduced our main enemies in acoustic design last month, it's now time to check out some of our allies. Absorbing mid and high frequencies is not much of a problem.There are several proprietary acoustic tiles, foams and heavy drapes that can effectively soak up frequencies above 300Hz. However, dealing with the bass end requires a more rigorous approach.
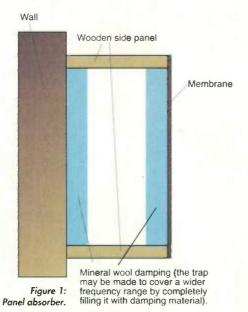
The reason for this is that low-frequency sounds have long wavelengths and a purely absorptive bass trap needs to be at least one eighth of a wavelength deep to do any good. At a frequency of 50Hz, that's approaching three feet and there aren't many studios that can afford the space to cover one or more walls with a three-foot thickness of mineral wool (such as Rockwool) with a density of 150 to 175 kg per cubic metre. That said, this type of trap has the advantage of working equally well at all frequencies down to its lower cut-off point.

The other, and understandably more popular, approach is to build a damped, resonant structure that will absorb a significant proportion of a specific frequency band by converting it to heat via frictional losses. As the sound energy expended by a Wembley football crowd during an FA cup final (including extra time!) would be barely sufficient to warm a pot of tea for the teams at half-time, you don't have to worry about thrash metal bands setting fire to your walls during over-zealous guitar solos!

RESONANT TRAPS

There are two commonly used traps, both of which are easy to build. They are the panel absorber and the Helmholtz resonator. However, to be successful you first have to know precisely where your problem frequencies are and then you have to build these traps accurately to ensure that they work at the right frequency. I've included them here more for academic interest, as in a nearfield monitoring situation you can probably solve your worst problems by less complex means.

Both types of absorber take up a large area, but have the advantage of being only a few inches deep. Even so, you must bear in mind that these are tuned traps and so are normally used to reduce specific resonances. They are not suitable for use as broad-band absorbers, with the exception of a panel trap constructed with a highly damped, limp membrane.

Wall

Wooden side panel

Membrane

Mineral wool damping (the trap may be made to cover a wider frequency range by completely filling it with damping material).

Figure 1: Panel absorber.

PANEL ABSORBER

The panel absorber is the easiest and most predictable bass trap to design and build. It consists of a simple wooden frame over which is fixed a thin, flexible panel such as plywood, hardboard, barrier mat or even roofing felt. Fibreglass or mineral wool is often fixed inside the frame to help damp the system by absorbing energy, since the more the trap is damped, the wider the frequency range over which it will work. The resonant frequency is a function of cavity depth and mass per square foot of the panel material, so it's easy to calculate the necessary dimensions using a simple formula (shown in the box elsewhere in this article).

The actual area doesn't make any significant difference to the operating frequency, but obviously the more you want to reduce the low-frequency reverberation time, the larger the area of panel you'll need in the room. To understand better the effect of a given area of absorber, consider that a perfectly efficient full-range trap would affect the sound in the same way as an open window of the same size (but obviously without the associated problems of sound leakage).

Filling the cavity with fibreglass or mineral wool tends to lower the resonant frequency by up to 50 per cent as well as doubling the effectiveness of the trap. It also lowers the Q of the trap so that it is effective over a wider frequency range. A typical panel-type trap is effective for frequencies around
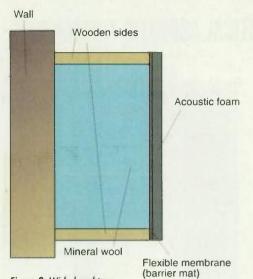
Figure 2: Wide-band trap.

one octave either side of the centre frequency, which at least has the advantage that you don't have to be absolutely accurate to get results.

As the surface of the panel may also reflect higher frequencies, curved panel traps have been constructed to simultaneously absorb bass frequencies and diffuse higher ones. However, it may be easier just to cover the front face of the trap with a layer of acoustic foam to extend its usefulness to the mid and high end of the audio spectrum. Figure 1 shows the construction details of a conventional panel absorber. Because the amount ▶

PRACTICAL ACOUSTIC TREATMENT

Wall

Wooden side to trap

Perforated
front panel

Figure 3:
Perforated
Helmholtz-type
absorber.

Mineral wool

of damping affects the tuning of the trap, you may want to test the finished trap to see what effect it's actually having. You can discover the resonant frequency by sticking a cheap contact mic on to the panel's surface, then plugging the mic into a preamp or mixer with a VU meter. Play a test tone from an oscillator or test tone CD using loudspeakers, and vary this around the frequency the trap is designed for until you get a maximum meter reading. This will be at the trap's resonant frequency.

DAMPING

An undamped panel trap using a rigid membrane will radiate some energy back into the room after the incident sound has ceased. This is clearly an undesirable state of affairs, and so some degree of damping is generally included within the panel. Though panel traps are normally considered to be tuned absorbers, the use of a heavy, well-damped panel material lowers the Q of the trap so much that it may be considered a broad-band device, especially when combined with plenty of internal damping.

There are specialist materials such as mineral-loaded vinyl barrier matting, or even lead-loaded materials, that are heavy, flexible and highly damped and therefore lend themselves to wide-band bass trap design. You can also get good results from a trap which is between eight and 12 inches deep, filled with mineral wool and covered with a simple roofing-felt membrane. The only

disadvantage is that the properties of roofing felt tend to change with age, so using barrier mat is probably a better long-term solution (if more costly). Apparently the BBC used to use roofing felt for their bass traps, but now they've replaced them with more up-to-date designs where the budget permits. Where the budget doesn't, their recommendation is to cover the fronts of the traps with carpet to reduce reflections and to increase low frequency absorption. I assume the carpet also acts as a membrane in conjunction with the roofing felt. You can also try experimenting with heavy vinyl floor covering as a membrane.

With such a high degree of damping, the action of the trap is less like a resonant panel and more like a floppy wall. In other words, sound energy is expended in trying to vibrate the felt which is so well damped that the energy is largely absorbed. Because the Q of such traps is low, the depth of the trap becomes far less critical. A broad-band trap is shown in Figure 2.

Many historic buildings feature wood-panelled rooms and these often have well-controlled acoustical properties because, in effect, a panelled wall with an air space behind acts as a tuned bass absorber. To a lesser extent, studio construction involving plasterboard fixed to a frame also acts as a trap for bass and mid frequencies, although the actual results depend on the depth of the air space behind it. In practical terms, it is easier to treat a room that has a lightweight construction than one that is solid, because a large proportion of the bass energy passes straight through the walls instead of being reflected back. Unfortunately, in this particular case, what is helpful for acoustic treatment is totally at odds with what is desirable for good sound isolation — that is unless you are dealing with a lightweight inner shell built within a solid outer shell. If you have plasterboard walls, you can again try the sweep oscillator and contact mic approach mentioned above to establish the actual resonant frequency. If the walls are too resonant or resonate at the wrong frequency, adding a second layer of plasterboard, ideally one with a different thickness to the first, will damp the resonances and change the frequency of absorption. You can use the first formula shown in the box to establish what the new resonant frequency is likely to be.

HELMHOLTZ TRAPS

Another type of tuned trap, popular in broadcast studios and older recording studios, is the Helmholtz resonator. This is essentially an enclosure with an aperture, not dissimilar in terms of its physics to a bottle — and just as a bottle has a very specific resonant frequency (which you can hear if you blow over the hole), so too does a Heimholtz resonator. A bottle has a very narrow bandwidth, but by introducing an absorbent material such as fibreglass or mineral wool into the neck to reduce the Q, the operating range can be widened. While you don't see many studios full of bottles (at least not ones being used as bass traps), it is possible to simulate the effect of hundreds of tuned bottles ▶

THOSE FORMULAS IN FULL

• **PANEL ABSORBER FREQUENCY FORMULA**
The formula for determining the frequency of a simple panel absorber is as follows:

$F = 170/\sqrt{MD}$

F = the frequency you're aiming to absorb

M = the mass of the panel in lbs per square foot

D = the depth of the air space in inches.

The metric equivalent of this equation is:

$F = 60/\sqrt{MD}$

where the mass is in kilograms and the unit of length is metres.

• **HELMHOLTZ TRAP FORMULA**
You can determine the operating frequency of a Helmholtz trap with the following formula:

$R = 200 \sqrt{P/DT}$

R = resonant frequency,

P = the percentage of perforation (total hole area divided by panel area multiplied by 100)

T = the effective hole depth in inches (thickness of panel plus 0.8 of the hole diameter)

D = the depth of the air space in inches.

Note that it's easy enough to convert this formula into metric, but the numbers are more straightforward if you stick to imperial measurements.

World Radio History

PRACTICAL ACOUSTIC TREATMENT

▶ with a perforated panel over an air space.

By fixing a perforated wooden panel over a frame and putting an absorbent material inside the space created, a resonant bass trap is formed with each perforation acting as a single 'bottle' in our virtual bottle array. Again, there is a fairly simple formula to determine the operating frequency (see box).

Figure 3 shows the construction of the Helmholtz trap. By varying the percentage of perforation, the design can be applied to both the bass and mid range. However, predicting the performance of these traps is difficult because the Q or bandwidth varies depending on the amount of internal damping.

The other problem is getting the right perforation percentage. Common pegboard is usually used in mid traps rather than bass traps. For example, pegboard with 3/16-inch holes on a one-inch matrix has a perforation percentage of 2.75 per cent. Fixed over a four-inch air gap, this gives a resonant frequency of a little over 400Hz. Boards of different perforation percentages may be available from specialist acoustic suppliers, but are not readily available from conventional builders merchants.

Like a panel absorber, adding an absorbent material lowers the resonant frequency slightly and also broadens the resonant peak. Instead of employing perforated board, it's theoretically possible to use a series of slats to create a slotted panel with the correct perforation percentage, but the calculations for such an absorber don't always predict a precise result, so I'd recommend some form of test to verify the final resonant frequency.

Helmholtz resonators were once very widely used both in broadcast and recording studios, but panel traps with limp membranes seem more widespread in modern designs.

Tuned traps should be placed on the walls corresponding to the room modes that you wish to attenuate. Your can tackle floor-to-ceiling modes by placing a trap on the ceiling. Traditionally, bass traps are placed close to corners where there is an area of high pressure.

MID AND HIGH ABSORBERS

One of the simplest absorbers for use at higher frequencies is open-cell foam, such as that used in furniture. In fact, expensive acoustic foam tiles are often only sculpted versions of this same material. (Note that for safety reasons, you should always use fire-retardant foam.)

The lowest frequency that will be effectively absorbed is dictated by the thickness of the foam. One-inch thick foam is most effective above 1kHz, while four-inch thick foam is useful down to 250Hz. The low-frequency absorption can also be improved by spacing the foam a few inches away from the wall using a wooden frame.

A similar absorber can be made from two-inch mineral wool slab, fixed to a frame two inches away from the wall and covered with open-weave fabric to prevent the fibres escaping into the air. Again, this should be usefully effective down to

250Hz or so. A variation on this is the acoustic blanket used in broadcast work where layers of mineral wool reinforced with lightweight wire mesh are covered with fabric and hung from walls. The frequency down to which a blanket is effective is determined by the airspace behind.

Carpet is not thick enough to be effective at anything more than the high frequency end of the spectrum — its absorbency drops off noticeably below 2kHz. There is a slight advantage in using a foam-backed carpet, and once again, mounting this with an airspace behind will extend its effectiveness down another octave or so. However, it's less than ideal as a wall treatment.

Variable absorbency in the mid- and high-frequency range can be achieved by hanging heavy drapes a few inches from the wall. The width should be generous enough to allow the material to hang in folds rather than being tightly stretched. If they are hung on a rail in front of a reflective surface, it's a simple matter to draw them back when you want to convert a dead acoustic into a live one.

SUMMARY

Absorbent traps can be useful in producing a more even acoustic environment, but only if applied intelligently so as to produce a nominally consistent reverberation time across the audio spectrum. The most common mistake people make when building their own studios is using too much trapping, usually at mid and high frequencies, and this serves only to further emphasise low frequency resonances that are more difficult to cure. In any event, it's probably unwise to do anything irreversible before the carpets and equipment are installed in the studio as these invariably make the room sound totally different to the way it did when empty.

Numerous advances in trap design have been made by specialists in the field of acoustic treatment, with the result that it's now possible to treat a completed room by adding just a few well-chosen panels in the right places. It's also possible to build panels with variable absorbency so that they can be adjusted in situ. Understandably, the designers of these traps are reluctant to give too much away! If you have a difficult room and you're planning to do commercial work, it could be cheaper to call in a proficient acoustic consultant than to tackle the job yourself and get it wrong.   **SOS**

The DDP takes the five dynamics processes you're most likely to want, turns them digital, and squashes them into a 1U studio workhorse box for under £600. HUGH ROBJOHNS wonders if it's all too good to be true...

Possibly the best known manufacturer of analogue dynamics processors — compressors, gates, de-essers and so on — dbx have employed the all-conquering binary bit in their first stand-alone digital dynamics machine, the DDP. The name is a contraction of Digital Dynamics Processor, and that's exactly what the unit is, so why not?
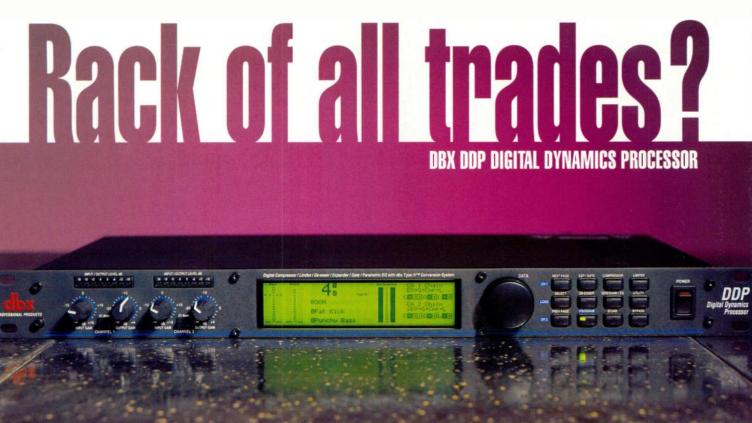
Rather than merely producing a digital replacement for a conventional compressor or gate, what dbx have created in the DDP is an amazingly powerful and compact multi-function processor, a machine which can perform up to five dynamics functions simultaneously on a stereo signal, or four processes on each of two independent mono signals.

The DDP offers a number of preset processing configurations: two stereo setups, six dual-mono configurations, or one single-channel mode with external side-chain keying. The sequential order of the dynamics processes can not be altered, but the options (in the prescribed input-to-output order) are:

level controls with associated bar-graph metering.

The rear panel is just as straightforward, with XLRs and TRS quarter-inch jacks provided for the electronically balanced analogue inputs and outputs. There's also the obligatory pair of MIDI In and Out/Thru sockets. An optional digital I/O module is available (but was not fitted to the review model), providing AES-EBU and S/PDIF interfaces at 24-bit resolution and 44.1 or 48kHz sample rates.

Analogue inputs and outputs are converted at 24-bit resolution with a quoted dynamic range of 105dB (A-weighted). However, dbx have incorporated a sophisticated soft-limiting function in the A-D stage, and this is claimed to increase the effective dynamic range significantly. The so-called Type IV Conversion System is supposed to prevent brief input overloads from transients turning nasty in that familiar way that digital equipment has, and, using this system with suitable transient-rich material, the specs claim that the usable dynamic

Rack of all trades?

DBX DDP DIGITAL DYNAMICS PROCESSOR

- 3-band parametric equaliser (inserted in the side-chain in some configurations)
- Expander/gate
- Compressor
- De-esser
- Limiter

There is also a sixth block, which uses channel 2's input to drive the side chain of channel 1, enabling external keying and ducking.

HARDWARE

The DDP is a typical 1U rackmounting device with the traditional black paint finish and a very readable yellow back-lit LCD window in the centre of the front panel. A large data-entry knob and function select buttons cover the right half of the panel, while the left features input and output

range is something more like 122dB (A-weighted).

Unusually, there is a built-in calibration routine to align the A-D converters, although the manual offers no suggestions as to how often this needs to be done, or what audible degradations may be observed when the converters require re-calibrating! However, during the time I had the unit I didn't notice any problems.

I'm pleased to report that the machine has an internal mains power supply with an IEC socket fitted at the rear and a front-panel power switch. There is no provision to change the mains voltage, nor externally accessible fuses, and I was surprised at how hot the casing around the power supply area becomes after a short period of use. It certainly was too hot to rest a hand on, although the machine didn't seem to suffer! I would suggest

that if the DDP is destined for rackmounting there should be at least 1U of space above and below, to allow air to circulate freely.

CONTROLS

The DDP's input and output gain controls are detented and are calibrated from minus infinity to +16dBu and +4dBu respectively. The 0dBu marks are at different physical positions on the scale, giving an odd visual impression — the input zero point is at 12 o'clock whereas the output zero is at 3 o'clock.

An illuminated push button between each pair of gain knobs assigns the relevant LED bar-graph meter to showing either input or output levels. However, there are no markings to say which position of the button is which (for the record, the button illuminates bright green when the meter is showing the input level). The front-panel meters show signals down to -24, with the usual traffic-light arrangement of coloured LEDs, but there are also two pairs of bar-graph meters on the left of the LCD screen, showing the input and output levels down to -50dBu. These meters have a further advantage, in that they indicate both peak and average levels simultaneously through the use of a fixed column with a floating bar.

The right-hand side of the LCD is taken up with the 'curve window' — a graphical display showing either the composite input/output transfer curve (for all the dynamics processes simultaneously); a frequency response plot when the EQ section is recalled for adjustment; the frequency selectivity for the de-esser; or the processing 'chain' structure, together with a set of threshold display boxes (more on this in a moment). To the immediate left of the curve window a pair of gain-reduction meters show the total amount of gain reduction applied by all processes, and these remain active even when the unit is bypassed.

Clearly, with up to five different dynamics processes going on at the same time, it can be hard to figure out just which element is doing how much of what! The DDP provides a collection of small graphical boxes in the top-left corner of the curve window, or on the main 'chain' menu page. The compressor is allocated a row of three of these little 'threshold meter' boxes, marked '-', '0' and '+'. At any one time, one box is reversed in colour from the others, indicating whether the signal is below, at, or above the threshold. The gate and limiter have similar facilities, but are restricted to just '-' and '+' boxes; the de-esser has a lone '+' box. This novel system gives the user an at-a-glance idea of just what is going on, although I found it took a bit of getting used to.

DYNAMICS PROCESSING

All of the DDP's pre-programmed effects 'chains' include a compressor as the central element in the signal path, with variations over the surrounding dynamics modules. Most chains add a gate at the front of the signal path and a limiter at the back, while others introduce a de-esser after the compressor, or an equaliser before the gate. In several of the modes the equaliser is moved to the side-chain, so that a spectrally modified signal controls the dynamics processes. A point worth making here is that all the dynamics processes share a common side-chain signal derived from the input. This means that adjusting one processor does not interfere with the performance of the others — something which can be advantageous over the more usual arrangement of daisy-chaining separate analogue units.

Setting up the machine and navigating the menus is, frankly, tedious, frustrating, and not good for the band waiting in the studio! The basic problem with the DDP is that there are not enough real-time controllers (knobs to you!) and the LCD only shows three settings at a time for each selected process. With the compressor, for example, that means three pages of menus to trawl through. This wouldn't be so bad if the really important parameters were gathered on the first page of each set of menus — but they're not!

In my opinion, the three most crucial parameters of a compressor (or most other dynamics processors, for that matter) are the ones which have to be adjusted all the time to get the best from the machine. These include the on/off function, the threshold, and the release or recovery time. However, with software v1.110 on the DDP, the proffered parameters on the first menu page are on/off, knee and auto mode on/off (automatic signal-related attack and release times). The threshold control turns up on page two and the release time on page three, requiring two and four button presses respectively to allocate the data-entry wheel to them.
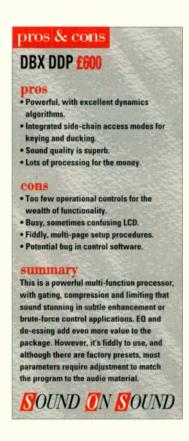
In a busy recording session you need to be able to reach for the appropriate knob and tweak the setting immediately. By the time I had found the right parameter in the DDP, the singer had usually finished the verse, and although familiarity increases the speed of operation, this mode of parameter access is never going to be as quick or obvious as reaching for a dedicated knob. I would have preferred to see a couple of data wheels defaulting to threshold and recovery time, with a push-button to bypass the dynamics module.

There's also a possible bug in the software concerning some inconsistency in how the front-panel controls are handled. A slightly lingering touch on the Limiter or Expander/Gate processor buttons causes the Next Page button to illuminate and the menu pages to cycle around at break-neck speed. Pressing and holding either of the Compressor or De-Esser buttons did not have the same effect.

DYNAMIC TOOL BOXES

The inclusion of 50 factory programs designed to lend the 'LA vocal' sound or a 'radio mix' feel to the audio signal is an interesting concept. The hardware makes it easy to do, of course, but its value is debatable. Providing a number of different module configurations is totally valid and worthwhile, but I remain doubtful about the practical use of many of the programmed settings, other than as a starting point. Small variations in input level can have enormous effects on the quality and character of the processed sound and I found I couldn't use factory presets in their raw state, but always had to tweak and tune virtually every parameter, to optimise them for specific audio material.

The various dynamics processors all have as comprehensive a range of controls and parameters as could be wished for. They can all be switched on

"All of the dynamics processes sound superb and work extremely well — even when pushed pretty hard."

DBX DDP

> "The basic problem with the DDP is that there are not enough real-time controllers (knobs to you!)."

and off individually, and have the usual threshold, ratio, attack and release settings. The last is interesting: dbx have chosen to calibrate the release time as the rate at which gain returns to normal, rather than the approximate time it takes. Although the former is, strictly, the correct way to calibrate a release time parameter, there are few other machines on the market scaled in this way, and I would suggest that few people will feel comfortable when confronted with a range of settings between 360 and 5dB/sec! In practice, I'd hope the ears would be the final arbiter of a suitable release time setting, but those who like to 'paint by numbers' may find this aspect of the DDP a challenge!

The gate module adds to the basic selection of parameters with a hold facility (setting the time for which the gate remains open after the signal has fallen below the threshold), and a 'Transient Capture Mode'. This TCM system inserts a user-selectable delay (up to 3ms) in the signal path, allowing the side-chain to take a sneak preview of the audio signal. The idea is to allow the gate to be opened fractionally in advance of an arriving transient leading the wanted signal. Most conventional gates can't open until the transient has exceeded the threshold, and thus part of the wanted signal's transient is often missed. I found the TCM system worked extremely well; the quality of a heavily gated signal was remarkably good, and far better than many popular analogue gates.

The compressor processor supplements the basic collection of operational parameters with an automatic signal-related attack and recovery time setting, make-up gain (of +/-20dB), hold, and the inevitable (since this is a dbx product) OverEasy soft-knee facility, complete with 10 different transition slopes.

The de-esser has only three parameters — on/off, frequency selection, and de-essing amount — while the equaliser is a full-facility, 3-band parametric design. Centre frequencies of all three bell-shaped sections can range between 25Hz and 20kHz, with Q values of 0.25 (wide) to 16 (very narrow), and cut or boost of up to 12dB. If the equaliser is used in a program which places it in the side-chain, a monitor mode option becomes available, allowing the side-chain signal to be auditioned while the EQ is adjusted.

SQUEEZING THE PIPS

The DDP does exactly what it is supposed to do, in every sense. All of the dynamics processes sound superb and work extremely well — even when pushed pretty hard. For example, if correctly set up, the DDP's compressor causes none of the usual dulling which tends to beset analogue compressors when they are pushed hard. The sets of operational

controls provided for each process are exactly what are required, allowing precise adjustment of every aspect of the machine and its treatment of audio signals. However, there are a number of other aspects of the DDP which do not impress as much. For example, I feel that using the relatively uncommon dB/second scaling for release time constants is a big mistake. Virtually every other compressor and gate on the market specifies release time as exactly that — a time for the gain reduction to return to unity — and the DDP's settings caused considerable confusion and debate amongst other engineers.

A rather more serious issue is that adjusting thresholds and release times for the various processes quickly becomes extremely tedious, frustrating and just plain slow! Having identified a process which requires adjustment, you have to select it, scroll around the various menu pages to find the one containing the desired parameter, assign the parameter to the data wheel and, finally, make the adjustment. Then you have to repeat the procedure all over again for the next process! The same rigmarole applies should you wish to bypass a single dynamics process to check on its contribution.
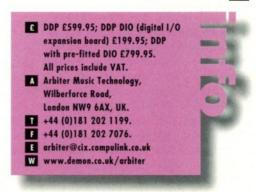
In a multi-function machine, with a limited number of controls, this kind of ergonomic and operational problem is to be expected, I'm afraid. However, some manufacturers find practical systems which can be used effectively in the heat of a real studio situation rather than the R&D lab. Given the choice, I would still prefer to stick with a rack of separate outboard processors, simply for the benefits of dedicated control knobs.

I really feel that the DDP should have had at least a couple of knobs which defaulted to the threshold and release times of any selected process, since these are the two most critical controls for any dynamics unit. In conjunction with the dedicated buttons to select a wanted process (perhaps with a double-press facility to bypass individual sections) this would make the unit a whole lot more usable in a real working environment. I know the provision of more knobs would add to the cost of the machine, but its excellent sound quality easily justifies a little more than its very modest current price tag anyway.

Although the DDP packs an awful lot of dynamics processing power into a compact box, and is capable of stunningly good results, setting it up is certainly not a trivial business, despite the 50 preset programs. Rather like a child prodigy, the dbx DDP is gifted, but difficult to deal with. **SOS**

A BIT OF TSE

One important function associated with the DDP's equaliser is 'Tape Saturation Emulation'. There are five TSE settings: dark, warm, none, light and bright. This feature is implemented through the A-D converter circuitry and is therefore not available to digital inputs — a shame, since digital sources are more likely to benefit from such a process. The TSE effect is quite subtle and really only affects high-level signals, imparting the kind of smoothness associated with analogue tape saturation in degrees described very well by the names of the settings.

£ DDP £599.95; DDP DIO (digital I/O expansion board) £199.95; DDP with pre-fitted DIO £799.95. All prices include VAT.

A Arbiter Music Technology, Wilberforce Road, London NW9 6AX, UK.

T +44 (0)181 202 1199.

F +44 (0)181 202 7076.

E arbiter@cix.compulink.co.uk

W www.demon.co.uk/arbiter

ARE YOU AWARE?

EURODESK

There are still people who claim that a good, professional mixing console costs a fortune. BEHRINGER's EURODESK series once again proves them wrong. The MX8000 has already received rave reviews for its dynamics, audio transparency and versatility. Now let your creativity run wild with the MX2442, while still keeping full control. Eight busses, six auxes and a full featured Mix-B section in the MX8000 or four busses and six auxes in the MX2442 give you control and flexibility, whether live or in recording. Our robust 19" rack mounted power supply units and the finest components guarantee an exceptional and reliable performance.

BEHRINGER

World Radio History

One bit at a time

ALL ABOUT DIGITAL AUDIO

PART 4: In the fourth instalment of our series on the techniques and technology behind digital audio, HUGH ROBJOHNS looks at digital tape recording formats.

In the previous parts of this series we have examined the essential building blocks of a digital system: sampling, quantising and error correction. This month, we look at some of the practicalities of recording and replaying digital audio data streams, starting with digital tape formats.

The first decision facing the designer of a digital tape recorder is whether to employ a stationary-head transport (like a conventional analogue recorder), or a rotary-head one (like a video machine). The former is mechanically simple and therefore relatively cheap, but has a low head-to-tape speed that means either a low data transfer rate, or a very high density of data on the tape (or both!). Rotary-head machines have high head-to-tape speeds and therefore high data transfer rates, with lower data density on tape reducing the demands on the tape medium itself.

In the early days of digital recording, it was hard enough just getting the A-D and D-A stages to work properly, so it made sense to redeploy existing video recorder technology rather than develop a bespoke digital recorder from scratch. Thus, the professional CD mastering recorders used off-the-shelf video transports (professional three-quarter-inch U-matic machines) to store digital audio data encoded as black and white dots in a standard video picture format.

However, there are a few digital tape systems that have employed the stationary-head concept — most notably the Sony and Studer DASH-format multitrack recorders (more on this later in this article). Mitsubishi also produced an excellent 32-track stationary-head recorder for a while, but ceased production several years ago. Another stationary-head format that can be found occasionally (often in the reduced section of High Street hi-fi outlets) is the Philips DCC — Digital Compact Cassette (see the box elsewhere in this article for more on this format).

ROTARY-HEAD SYSTEMS

In a rotary-head recorder, the tape is partially wrapped around a rotating drum that contains a number of recording/replay heads. As the drum is angled very slightly off-vertical, the tape spirals around it and the heads trace a shallow diagonal stripe across its width. The drum rotates very quickly — around 1500rpm in the case of a video machine — which means that the head-to-tape speed and the data transfer rate is very high. However, the linear speed of the tape is relatively slow so that the narrow recorded stripes lie parallel to each other in turn making for economical usage of tape. See Figure 1 for a typical video head-drum arrangement.

This approach works very well, but using video machines to record digital data is a case of serious over-engineering! For a start, a video signal needs a signal-to-noise ratio of around 30dB, whereas a digital system only needs about 10dB. Second, the tape has to wrap completely around a video head-drum to allow continuous recording whereas digital signals are inherently discontinuous. Thus, the tape loading and wrapping mechanisms can be much simpler. Third, because video systems record audio as a linear track, the transports have to be built to minimise wow and flutter — an irrelevance in a digital recorder. These 'refinements' (together with other simplifications described below) allow a bespoke rotary-head digital recorder to be built more cheaply than a conventional video transport.

DIGITAL AUDIO TAPE

DAT, which appeared in 1987, was one of the first systems to use dedicated rotary-head digital recorders. The most obvious difference between DAT and video transports is that the tape wrap around the head-drum is only 90 degrees, as opposed to the 270 degrees of a video mechanism (see Figure 2). Not only does this considerably simplify the lacing procedure, it also allows the tape to be spooled against the heads with minimal friction — a serious problem with the longer wrap of video machines. Also, the fixed heads in a video recorder, which are needed for full tape ▶
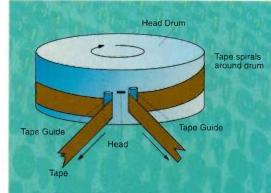
Figure 1: *In a video machine, the tape is arranged to wrap around the head drum for about 270 degrees of its circumference to ensure there is always at least one head in contact with the tape for a continuous recording.*

Figure 2: The tape and transport arrangement of a typical DAT machine.

Figure 3: The DAT tape track format.

erasure, audio recording and control tracks (see below) are not required in a digital machine, saving more costs. But the head drum still contains at least two heads that take turns to record digital data whenever they come into contact with the tape (for a quarter of each drum revolution).

However, not having static heads does introduce significant problems. Video machines align their replay heads with the recorded tracks on the tape through two means: a 'control track' which marks the start of each stripe, and a 'guard band' of erased, empty tape which separates adjacent stripes to minimise crosstalk between them. The control track makes sure that each head starts in the right place, but if the tape speed is slightly wrong the head will wander off course and the signal level will fall as it enters the guard band. Thus, the signal level is used as a measure of the correct tape speed and head tracking.

DAT machines don't have control track or erase heads, and there is no guarantee that the tape will be fully erased (if an old tape is being re-used, for example). Erasure is not a prerequisite for digital tape recording as the magnetic medium is fully saturated as North-South or South-North magnetic elements during recording — a new recording simply overwrites the old one. However, this means that there won't be empty guard bands between tracks, so some other method of locating and maintaining the replay heads on the correct tracks is necessary.

A relatively wide track of about 20µm (micrometres) in width and 24mm length is laid down across the tape during recording by the first head in the drum. As the second head comes around, it then records its stripe to partially overlap the first one, guaranteeing every part of the tape is recorded and that no previous recording survives. The data stripe left on the tape ends up being about 13.6µm wide (see Figure 3).

Since the record/replay heads are around 20µm wide, but the recorded stripe is only 13.6µm wide, the heads will obviously overlap adjacent tracks during replay, potentially leading to considerable crosstalk. However, a clever technique called Azimuth Recording is used to overcome this. One head is twisted 20 degrees to the left and the other 20 degrees to the right creating a 40-degree azimuth error between them — and also between alternate recorded tracks. Although each head can replay its own recorded stripe, it picks up very little crosstalk from the adjacent tracks, obviating the need for guard bands and allowing much greater packing density on the tape. However, a mechanism is still required to align the heads in the right place to start with.
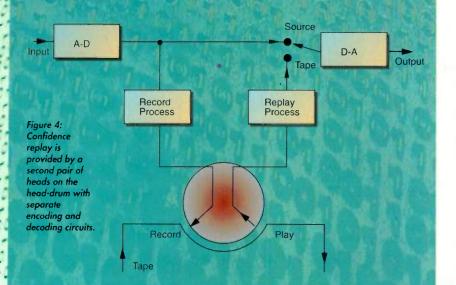
Not all of what goes on to tape when you make a digital recording is digital audio — various other bits of data are recorded too. The stereo audio data is recorded in the central portion of the stripe and is bounded on each side by a sequence of tones called ATF (Automatic Track Following). Subcode data is also recorded between the ATF and the outer edges of the stripe (more on this in the 'Subcodes' box).

ATF signals comprise a repeating sequence of relatively low-frequency tones, with the lowest at 130kHz. Such tones are not affected by the severe head azimuths, and remain equally detectable to both heads. The tone cycle repeats every four recorded tracks (in other words, every two revolutions of the head-drum) and the sequence is used to align the relevant head with the appropriate tape stripe, acting in much the same way as a video machine's control track. ATF signals are found on both sides of the audio data in the stripe, so should the head wander off course, they help to control the tracking.

The use of azimuth recording, overlapping tracks ▶

ADAT & DTRS

The DTRS tape format used in the Tascam DA38/88/98 modular digital multitracks (as well as the Sony PCM800) is essentially an overgrown DAT tape. It uses Hi-8 video tape as the medium and most of the physical dimensions are scaled up, but otherwise it is fundamentally the same as a four-head DAT machine configured for pre-reading. Like most digital recorders, only one data stripe is recorded by each head in the head-drum and the eight audio channels (plus timecode) are all multiplexed within that data stream. Consequently, tapes must be 'formatted' before use (or all eight tracks recorded at once) so that the multiplexing structure is complete on the tape.

Overdubbing involves retrieving the multiplexed data stream from the tape with the pre-read heads, de-multiplexing and error-correcting to produce eight discrete sequential channels. One or more channels can then be replaced with new audio material before re-encoding, multiplexing and recording the entire re-combined data stream back on to the tape in the same physical position as it started.

The ADAT format is much the same, but adds analogue auxiliary and timecode tracks. It also employs the much larger S-VHS medium. DTRS seems to have become the de facto standard in television and film dubbing, as well as in music-for-film applications. However, the ADAT is the undisputed champion of many professional, semi-pro and home music studios. DTRS is restricted to 16 bits per channel (although bit-splitting boxes are available to allow a smaller number of higher-resolution channels), whereas the latest generation of the ADAT family (the new M20, for example, which will be reviewed in SOS shortly) provides 20-bit per channel recording.

Improved

Vocal Range

Continuing the tradition of innovation which brought you the ground-breaking Vocalist Performer, DigiTech have now enhanced their Studio Vocalist and Vocalist Workstation. The new-look 'EX' versions have MORE features and NEW presets. There is also the introduction of the NEW 1u rackmount VOCALIST ACCESS to complement the range.

VOCALIST PERFORMER

Incredibly life-like three-part harmonies within the range of *every* performer. The most affordable harmoniser yet - and so easy to use! Just hit the key and scale buttons and you're free to concentrate on *your* performance!

- Up To 3 Part Harmony (Original + 2 Voices) Of Stunning Quality
  - Auto Doubling Effect
  - 50 User Presets With 6 Style Variations
  - XLR Mic Input/ Stereo Outputs
  - Digital Reverb
  - A/B Part Programming (For Bridge, Chorus Etc.)
  - Optional Footswitch For A/B Switching, Reverb and Bypass
  - Ideal For Live Use

NEW *Vocalist Products From DigiTech...*

Includes FREE Mic.Stand Mounting Kit

VOCALIST WORKSTATION EX
- Up to 5 part intelligent harmony • Chordal, Scalic and Vocoder Modes • Gender Bender'© • Digital Reverb • 50 User/50 Factory Presets • MIDI Control • Pitch Correction • Automation Ready via Cakewalk Studioware Panel • NEW PRESETS

STUDIO VOCALIST EX
- High Quality Mic Pre-amplifier • 18 bit A/D Converter • 48kHz Sampling • 5 XLR and 1/4" Outputs • Automation Ready via Cakewalk Studioware Panel • NEW PRESETS

VOCALIST ACCESS
- Up to 5 part intelligent harmony with REVERB • Chordal, Scalic and Vocoder Modes • 48v Phantom Power • Comprehensive MIDI Control • Pitch Correction • Separate Outs For Each Voice • Aux reverb input • Specifically designed for keyboard & MIDI file users

DigiTech

Distributed exclusively by
ARBITER GROUP PLC
Wilberforce Rd. London NW9 6AX
Tel: 0181 202 1199 Fax: 0181 202 7076
Web: www.arbitergroup.com

World Radio History

Figure 4: Confidence replay is provided by a second pair of heads on the head-drum with separate encoding and decoding circuits.

"...most problems originate in the A-D conversion."

Figure 5: Glitch-free drop-ins can be made if the DAT machine has four heads to allow pre-reading of data from the tape.

that the machine is really recording. Off-tape monitoring is achieved by a second pair of heads in the head drum which are staggered in height relative to the first pair, typically by the equivalent of six tracks (three drum revolutions). In these 'four-head' machines, the data to be recorded passes through the record circuits, which add the error protection, interleave the data (see last month's part for more on this), and sort out the channel-coding structure, before being committed to tape (see Figure 4). This process typically takes about 45 milliseconds.

The replay circuits take about the same time to decode the data, perform the necessary error correction and so forth. The six-track stagger in the record and replay heads adds a further 90 milliseconds, so the total delay between input and off-tape monitoring is a 180 milliseconds — enough to give that familiar double

and ATF sequences increases the recorded data density on the tape considerably, allowing more than two hours of digital data to be stored on a relatively short length of tape. The success of these techniques has led to their becoming common in several other digital audio and video recorder formats.

OFF-TAPE MONITORING & EDITING

In analogue tape recorders, off-tape monitoring is essential, as it is the head/tape boundary which causes most audio quality problems — dirty heads, incorrect bias and so on. Listening to off-tape playback is therefore the only way of knowing whether the recording is OK. However, in digital recorders, the tape simply stores data and, major dropouts excepted, the tape is unlikely to be the cause of quality problems. In fact, most problems originate in the A-D conversion, such as signal levels being too low or too high. As all DAT machines provide monitoring through the complete A-D/D-A chain, true quality monitoring is always provided anyway.

Nevertheless, one of the biggest operational problems with a DAT recorder is knowing when it is really recording, simply because you can't see the tape going around very easily! Consequently, off-tape monitoring is provided on some professional and semi-pro models to give the operator the confidence

beat when switching between source and tape!

Last month, I mentioned the problem of dropping in to record on a digital recorder because of the disruption to the interleaving structure. One convenient way around this problem is to make use of the four heads in the drum, but in the reverse order — that is, employ replay ahead of recording (see Figure 5). In this case, data can be retrieved from the tape, decoded, de-interleaved and error-corrected, before being presented to a digital crossfader in normal sequential form. Initially, the crossfader passes the original data straight back to the tape through the record circuits which re-encode and interleave it.

Since the replay and record circuits each take 45 milliseconds to process the data, and the timing between the heads amounts to 90 milliseconds, the data should arrive back on the tape in exactly the same place as it started. The record head can thus be activated at the start of any stripe without regard to the interleave structure. The crossfader allows new audio material to be punched in and out smoothly without gaps or glitches. When the original data is being recorded back on to the tape over itself, the record head can be deactivated — a very elegant solution!
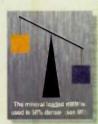
Most four-head machines allow the heads to be configured either for confidence replay or for gapless drop-ins but, in theory, there is no reason why a ►

▶ machine should not have six heads, providing both pre-reading and confidence replay simultaneously.

Many of the concepts and technology developed for DAT are used in other rotary-head recording formats — two more are described in the 'ADAT & DTRS' and 'Nagra D' boxes elsewhere in this article. Due to the similarities between DAT and these other formats, the boxes highlight only the major differences.

STATIONARY-HEAD FORMATS

It might seem that rotary-head recorders rule the roost as far as digital tape recording is concerned, but there is a finite limit to the maximum data transfer rate that can be achieved, and until recently, it was not sufficient for a 24- or 48-track recorder. However, stationary-head machines with suitable tape moving quickly can be designed with sufficient transfer rates, and these have therefore become the dominant format for 24 and 48-track applications.

There are only three common stationary-head digital tape formats: PD, DASH, and DCC (see the separate box for more on DCC) — and only one of those is likely to be around into the next millennium. Pro-Digi (PD) was Mitsubishi's digital multitrack format — a range of machines offering 2, 16 and 32-tracks on quarter-inch, half-inch and 1-inch tapes respectively. The 32-track machine uses 10 physical tracks for every eight audio channels (the two extra tracks carried some of the error protection data). Thus there is a total of 40 tracks for the
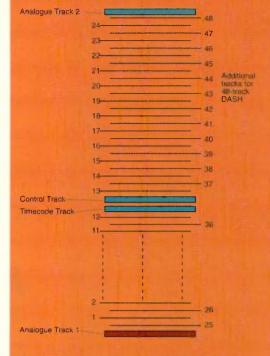
Figure 6: The 48-track DASH tape track layout. Studer have also joined Sony in manufacturing DASH machines.

digital audio, plus two auxiliary data tracks, two cue tracks and a timecode track — 45 tracks in all!

Although these 32-track machines are still widely used, particularly in the US, they are no longer being manufactured: Sony's dominance of the digital multitrack market prompted Mitsubishi's early withdrawal.

Sony's DASH-format machines (Digital Audio Stationary Head) have become standard in many professional studios and are available in 2-track, 24-track and 48-track versions. The first machines appeared in 1987 with a 2-track using quarter-inch tape and 24-tracks on half-inch, unfeasible though this may sound! Two-track machines are not very common, but one of their marketing strengths is that they can cope with physical tape editing by way of razor blades and sticky tape, thanks to an elaborate interleaving structure. Provided that edits are at least 1.5 inches apart, the error correction can rebuild enough of the missing data to allow a smooth crossfade over the edit.

The earliest DASH machines used conventional ferrite head assemblies whose physical size limited the maximum number of tracks to 24. However, the newer machines use 'thin-film' heads (made in the same way as integrated circuits) which has allowed the introduction of 48-track machines. The standard multitrack DASH manages to record 24 digital tracks, two analogue guide tracks, a timecode track and a control track on half-inch tape running at 30ips. The analogue cue tracks are on the outer edges of the tape where they protect the delicate digital tracks, and the timecode and control tracks are in the most stable region of the tape in the centre. The 48-track version slots an additional 24 tracks in between the original 24 (see Figure 6). The latest versions of the machine also allow 24-bit recording through an built-in bit-splitting system that shares the 24-bit audio data channels across multiple tracks. **SOS**

DIGITAL COMPACT CASSETTE

Digital Compact Cassette (DCC) is a domestic stereo format intended to replace the analogue compact cassette, based on a stationary-head recorder. The basic idea was conceived at the same time as DAT in the mid-'80s, when the two formats were referred to as R-DAT and S-DAT (Rotary head and Stationary head respectively). Sony's original S-DAT proposals and Philips' DCC differ only in the details.
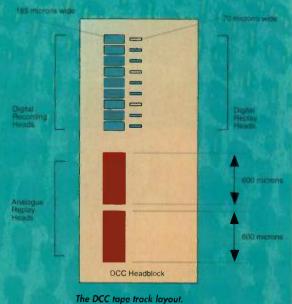
The tape speed is the same as that of an analogue cassette (1.875ips) which is far too slow to allow sufficient data transfer rates for a single track per audio channel. Instead, a 4:1 data reduction system is used (known as PASC — Precision Adaptive Sub-band Coding) to minimise the amount of data that is then shared between eight parallel tape tracks. A ninth digital track carries auxiliary and subcode information such as timings and track titles (see the diagram, right).

The DCC head block incorporates three sets of heads: nine 185μm wide digital record heads; nine 70μm-wide digital playback heads; and two 600μm-wide analogue playback heads. The recording heads are

more than twice the width of the playback heads so that alignment between machines is less critical, and the analogue heads allow standard compact cassettes to be replayed in the same machine.

Like the analogue cassette, the DCC is double-sided, but cannot be physically turned over. Instead, DCC machines either have two separate head assemblies, one for each side, or one assembly that inverts for side B.

DCC, although incorporating some very advanced technology, is unlikely to succeed in the domestic marketplace because the consumer has become used to the instant access afforded by CDs and Minidiscs. For some, DCC's credibility is also marred by the use of data reduction, despite the fact that it actually sounds extremely good.

The DCC tape track layout.

World Radio History

World Radio History

Sample shop

Our regular look at the hottest new sample CDs and CD-ROMs.

KOMPUTER INSIDE
(AUDIO CD)

As the '80s revival steadily gathers momentum [*The Wedding Singer, anyone? — Ed*] it was surely only a matter of time before someone allowed three avid electronic music fans free range over a studio filled with countless classic analogue synths from the early '80s and compiled the results in one single sample CD. The band responsible are an outfit called 'Komputer' and, as you perhaps might have guessed from the mildly cheesy mis-spelling of the CD title, the main influence behind their music, their band name and particularly their stylish tight-fitting black nylon polo-necks is, of course, Germanic techno pioneers Kraftwerk.

This 72-minute audio CD is a collection of riffs, single hits, effects, washes, drum loops, grooves and noises inspired and influenced by the earliest days of the modern synthesizer. The armoury of instruments behind this CD must be impressive, although we are left guessing as an instrument list is not included in the sleeve notes. The quality of *Komputer Inside* is actually very high; the sheer enjoyment the band must have had putting together the CD (totally obsessed with all things pre-MIDI as they are) is evident and infectious. The CD follows the familiar format of providing two- or four-bar grooves, then stripping down the constituent parts to allow more flexibility when sampling, and both the sound quality and the programmers' obvious mastery of their instruments makes for a highly usable and authentic trip back to the decade that brought us Rubik's Cubes, deelyboppers and 'mobile' phones the size of briefcases. Hardcore analogue freaks will surely appreciate the vast number of filter sweeps, bleeps, whizzes and whooshes, and the

selection of arpeggiated sequences is particularly impressive (if a touch too short). The drum sounds, despite being very authentic to the genre and mostly immaculately programmed, might sound a touch 'gentle' if compared to today's meaty dancefloor thuds, but the briefest of glances at the cover photo of the producers will immediately tell you that conformity is not high on this band's agenda, and this CD is all the better for it. It would be far too easy to say that only Kraftwerk fans are likely to be impressed by this CD, because I'm sure there must be a few Tangerine Dream fans still out there too, and they are going to love it just as much.

I'm personally not really into either band, but I had a load of fun sampling this CD. It's put together with integrity, artistry, a sense of humour and has a clear focus. Within its admittedly, narrow field *Komputer Inside* succeeds enormously. Not everyone will like it, and it would be fair to say that this is a specialist's product, but if you're looking for '80s retro with its tongue firmly in its cheek you would be foolish (or dare I say Krazy) to miss it. *Paul Farrer*
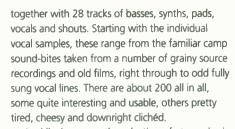
- **£** £59.95 including VAT and UK p&p.
- **A** SampleZone, Unit 2, Borehamwood Industrial Park, Rowley Lane, Borehamwood, Herts WD6 5PZ, UK.
- **T** +44 (0)800 731 2939.
- **F** +44 (0)181 207 4572.
- **E** sales@samplezone.co.uk
- **W** www.samplezone.co.uk

HOUSEMASTER
(AUDIO CD/ AKAI-FORMAT CD-ROM)

You could argue that the world needs another house music sample CD about as much as a motorcycle needs an onboard ashtray. Let's face it, there are hundreds of the things. While the sample CD market desperately busies itself trying to be as innovative as possible whilst trying to reinvent and redefine what a sample CD should be, however, Best Service have instead opted for a more 'meat and potatoes' approach and produced what could perhaps best be described as an extremely comprehensive sound history of the genre so far. *Housemaster* is a 72-minute audio CD (also available in Akai/Emu CD-ROM format) which offers the user over 1000 drum loops

together with 28 tracks of basses, synths, pads, vocals and shouts. Starting with the individual vocal samples, these range from the familiar camp sound-bites taken from a number of grainy source recordings and old films, right through to odd fully sung vocal lines. There are about 200 all in all, some quite interesting and usable, others pretty tired, cheesy and downright clichéd.

Luckily, however, the selection of atmospheric pads and hi-strings is fantastic, and all are totally authentic. Good use is made of the stereo soundfield, making for compelling sampling, and the collection of short 'house-synths' and organs that follow are equally good. The few single-sample bass sounds represent a fairly wide cross-section of the familiar noises heard on dancefloors across the globe, and what they lack in number they more than make up for in variety. The drum loops are cleverly divided into bpm groups each with a large number of deconstructed mix variants (with and without kick drum, snare, percussion, or whatever) and range from 124 to 138bpm. As far as the programming goes the loops (no longer than one bar each) represent a solid look back at how house music has developed over the past 10 years, rather than offering us any scorchingly original fresh new approaches. This seeming reluctance to forge any real new ground stylistically or sonically is both *Housemaster*'s great strength and simultaneously its main drawback. For any newcomer to serious sampling, I can think of no better product that gives you so much relevant and great material to get you started. Overall, though whilst it is totally focused on pure house, the truly professional programmer or DJ might well find a good many of these sounds a little too retrospective to consider them useful. Given that dance music (particularly house) looks here to stay, owning a CD like *Housemaster* will arm you with nearly everything you need to make a house track, imbued with all the bells and whistles (as well as clichés) that we have come to expect from snappy four-on-the-floor house tracks. Whether you can achieve anything truly fresh and innovative with this CD, as usual, depends on how creative you are prepared to be with the basic material. Perhaps that's what sample CDs should have been about all along. *Paul Farrer*
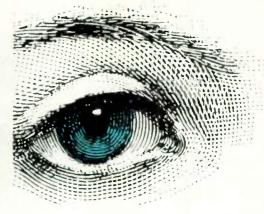
- **£** Audio CD £59.95; Akai-format CD-ROM £99.00. Prices include VAT & UK p&p.
- **A** Time & Space Distribution, PO Box 4, Okehampton, Devon EX20 2YL, UK.
- **T** +44 (0) 1837 841100.
- **F** +44 (0) 1837 840080.
- **E** sales@timespace.com
- **W** www.timespace.com

▶
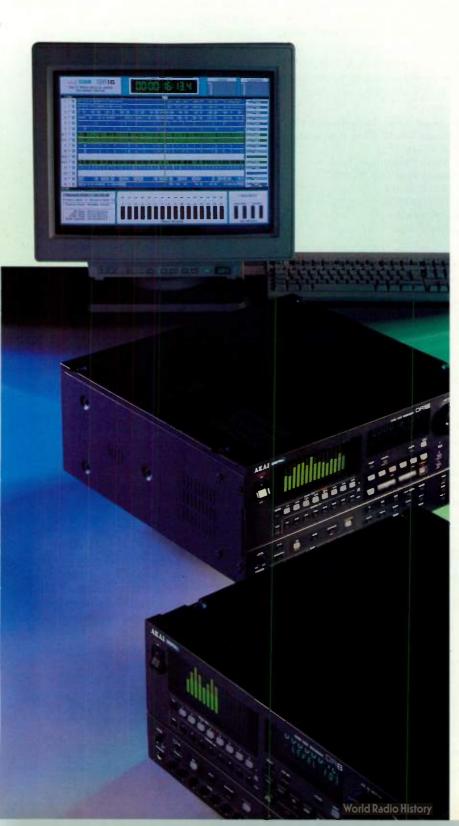
The **Akai DR** Hard Disk Recorders

Music to your **Eyes**

In the four years since Akai changed the future of recording with the 4 track DR4, the relentless pace of our development has produced the 8 track DR8 and the 16 track DR16 digital hard disk recorders fitted with custom designed LSI chips to allow highly sophisticated recording, mixing and EQ* performance in real-time. The magic of the DR Series is its high level of complexity and function, packaged to be as easy to use as a familiar, conventional MTR machine. And the DR system has a unique flexibility, it can be used as a dedicated, stand-alone product or can integrate easily with your existing system. It brings a highly cost-effective price level per track to high quality digital recording.

Now you have the option of the new IB807V graphics board for direct connection to an SVGA monitor, allowing the full 16/8 tracks of audio to be accessed and controlled on screen from a display based on the market-leading DD1500 post-production system. Edit, mix, record, playback and level are controlled simply and logically, with all tracks visible on screen, or a single track can be zoomed in for precise editing control. And all in real-time with no irritating screen re-draws, and no need for an external computer. We've added an ASCII input to the IB807V as well to let you work with a standard QWERTY keyboard as a full-function control interface with single keystroke shortcut facilities.

The Akai DR Series can grow from an 8 track upgrade for analogue MTR systems to a 128 track, multi-interface, multi-synched recording and post-production system with graphic based and remote operation and as cost is a factor we watch as closely as any other. Music to your pocket as well.

Akai is currently offering a special package which includes a DR16 fitted with 4 Gbyte hard drive, IB807V graphics board, S-VGA monitor and QWERTY keyboard. Imaginatively entitled **DR16VGA**, the package is available at **£3099 (inc VAT).**

Akai DR8 and DR16 Options

DL16 Remote controller.
DL1500 Multi machine editing remote.
EQ16 16 channel digital EQ. **EQ8** 8 channel digital EQ.
MT8 Mixer tab.   **ALX50** Remote cable.
IB807V Graphics board.   **IB801S** SCSI board.
IB802T SMPTE board.   **IB803M** MIDI board.
IB804A/AEX ADAT® board. **IB805R** RS422 board.
IB806B Bi-phase board.   **IB806G** GPIO board
IB809E Ethernet board (for use with DL1500)
IBD8MA AES/EBU board   **IBD8TIF** TDIF board

*Optional

AKAI DIGITAL

Akai (UK) Ltd
EMI Division, Haslemere Heathrow Estate
Parkway, Hounslow Middlesex TW4 6NQ
Tel 0181 897 6388 Fax 0181 759 8268
http://www.akai.com/akaipro

All products or company names are used for identification purposes only and may be the Trademarks of their respective owners.

sample shop

▶ **NO KICK**

(AKAI-, SAMPLECELL
MAC-, ROLAND-FORMAT
CD-ROM/AUDIO CD)

Everyone who deals in drumloops knows (or should know) about the problem of the bass drums in a loop flamming and phasing when layered with a programmed kick. XX Large, who are not noted for their inability to produce relevant and exciting dance sample CDs, have come up with the blindingly obvious solution to the problem; drum loops with no kick drums in them.

A pretty exciting demo outlines how a selection of these loops sound with and without kick. Then it's straight into the loops themselves. These are grouped by tempo, and tuned to sensible bpms (100, 124, 126, 128 and so on) — none of this fiddly 123.682 stuff. Every time you reach a new tempo, a Teutonic electronic voice intones the incoming bpm. The loops are huddled together, 20 to a track, and each track takes about a minute to run through. Most are pretty tuneless, so pitching them up or down a few per cents to sync with your song won't upset the basic feel or sound too much. In fact, quite high shifts are possible — remember, these loops ostensibly have *No Kick*, so virtually all the low energy in any kick/loop combination should be contained in your programmed kick — pitch the loop up and the bottom end of your rhythm track remains unchanged. Wonderful — in theory at least.

So what of the loops themselves, where do they come from, how were they made, how good do they sound? All loops are in stereo and are one bar long. Engineering is excellent throughout: silence reigns supreme between samples, all samples have similar energy with transparent (probably multi-band) compression adding punch with clarity, spacing between samples is consistent and picking off loops is easy. Nothing is repeated, though sometimes you get different versions of the same basic pattern. The vibe is up-for-it club rather than soul, and a mixture of programmed and live sounds is used. The programmed loops tend to be cleaner, and many use interesting electronic voices. The live drum loops (and there are many old favourites here) have either been digitally edited to remove the kicks, have never had kicks in the first place, or have had the kicks EQ'ed out. The latter method, which anyone with a decent high-pass filter could do for themselves, is used rather a lot on the second half of the CD, where too many kicks are more than barely discernible. Included in the pack are choppy, gated loops, loads of percussion, and some great filter sweeps. It would take me a week to evaluate all 1500 patterns in a looped situation, but on first and second listen I'd say there are plenty worth further investigation.

The title, by the way, applies to the loops, not the CD — there is a fine selection of kick drums at the end of this CD where the digital max tone should have been. With programmed kick and loop entirely separate, you are free to choose kick sound and pattern. You can, for example, filter sweep the loop while the kick holds steady. You can drop kicks at will. In fact, it will be a bind to go back to loops *with* kick after you have played around with *No Kick*. Now all we need is for somebody to bring out a similar product with soulful beats (hint).
Wilf Smarties

- **£** Akai-, Samplecell Mac-, Roland-format CD-ROMs £99.00 each; Audio CD £59.95. Prices include VAT and UK p&p.
- **A** Time & Space Distribution, PO Box 4, Okehampton, Devon EX20 2YL, UK.
- **T** +44 (0) 1837 841100.
- **F** +44 (0) 1837 840080.
- **E** sales@timespace.com
- **W** www.timespace.com

SMOKIN' BEATS WITH ATTITUDE

(AUDIO CD)

This is a very different kettle of drums from *No Kick*. A full kit is featured here, ably played in various traditional rare-groove-esque R&B styles by one Tony Mason. You know the drumming on the Paul Simon classic 'Fifty Ways to Leave your Lover'?. Think of the sound. Think of the dry but full snare. Think of the delicate snare work. That could've been a loop off this CD. In fact, the snare is what makes Tony Mason's drumming. Hi-hats tick along, the kick seems to find difficulty keeping club-tight on anything but a good solid R&B beat. But the snare...

First up, the sound of the main snare is lush. Perfect. Every frequency in exactly the right measure. Tight without being dead. Crisp

without being light. Tonally it has, in fact, much in common with a 909 snare, but the fidelity leaves the 909 standing. This is as good as 16-bit audio gets. The other snare Tony sometimes uses is tighter, tuned higher, but still with plenty of bandwidth. It's not quite so powerful as the first, however, with the compensating compression much more obvious, giving it a pseudo-non-linear (reverb) sound.

The patterns here are one or two bars long, repeated four or eight times with either the subtle variations that live playing imbues, or with conscious pattern variations. Beats are pretty straightahead R&B backbeats, with bpms (noted on the sleeve) that hover hover around the 100 mark, and which are sometimes made more interesting and upfront-sounding by some tidy snare work, drop shots, press rolls and the like. Where stereo is used it is done very subtly. Snare, kick and hi-hat reside pretty much bang in the middle, with ambient overhead mics (crossed or Middle-and-Side pair) bringing in a bit of width where required. The warmth of the close mics suggests valves to me. The sound of the kit is vintage, even

down to the squeaky hi-hat pedal. Engineering is generally excellent, though I did notice a pattern on track 5 where it sounded as though Dolby had been left un-decoded by mistake.

This is altogether a very human, old-fashioned set. All you hear (apart from the very occasional tambourine) is a drum kit. I'd hesitate to call the playing basic, particulary given the dexterity of some of the snare work, but Tony is definitely a team player, giving you a good, solid foundation on which to build a rhythm track. There are about 500 patterns in total of up to eight bars in length, which represents pretty good value. The style and sound of this CD are timeless, and only the fact that there are no fills or single hits to back up the beats might put you off. *Wilf Smarties*

- **£** £59.95 including VAT and UK p&p.
- **A** SampleZone, Unit 2, Borehamwood Industrial Park, Rowley Lane, Borehamwood, Herts WD6 5PZ, UK.
- **T** +44 (0)800 731 2939.
- **F** +44 (0)181 207 4572.
- **E** sales@samplezone.co.uk
- **W** www.samplezone.co.uk

STAR '80s INNOVATIONS

SOUND ON SOUND (NATCH)

THE ATARI ST

THE SONY WALKMAN

HOWARD'S WAY

CAGNEY & LACEY

World Radio History

Apple NOTES

Here's what's inside the 1.7Mb of Retro AS1 demo.

MARTIN RUSS looks forwards and backwards this month, with visits to the limits of Mac-Time, the year 2000, MacOS 10, and a modern version of yesterday's analogues.

Put on your brightest grin. Go on, let your face light up with glee. Because if you're reading this you've probably got a MacOS-based computer, and the next two years are going to be great fun.

When the 31st of December 1999 becomes the 1st of January 2000, something very interesting happens. Yep — it's the Millenium, that event which happens every thousand years when the year number acquires three zeros. But it's also the time when many computers may well become temporarily confused — and it's making some people very worried. It's also about to make some people very rich. It's the Millenium Bug.

2000 AND ALL THAT

The problem has to do with how some computers deal with years. Since 1900, all that has been needed to describe the year has been the last two digits: 98 for the current year, for example. But the year 2000 upsets this neat shorthand, since that looks like the year 1900. This apparently minor problem can be fixed by using more digits, but many computers don't at the moment. The consequences can be dreadful, and if you're selling software to fix these consequences you make them sound as dire as possible! But for most people, January 2000 will be filled with resetting dates, and

probably some other minor inconveniences, rather than the apocalypse than some commentators have predicted.

For us Mac users, it's very different. The Millenium has no effect on computers using MacOS, and neither does the dawn of the 21st Century a year later. In fact, Macs can cope with year dates from 30,081 BC up to 29,940 AD. I'm reasonably certain that this capability is unlikely to be required in 27 thousand years time, because I suspect that computers may be slightly different by then! While we're waiting, you can watch non-MacOS computer owners worrying.

BACK TO THE FUTURE

It's good to see a Mac software synthesizer whose programmers are working on eventually

sharing it with Windows owners — especially given the "Windows-only, no Mac version planned..." response of some companies. The *Retro AS1* from Californian programming house Bitheadz is a Power Mac analogue synthesizer emulation that incorporates OMS and FreeMIDI compatibility, 16-bit, 44.1kHz quality output, programming and performance controls, and more, into software — the Mac is your hardware. The program's name is a bit of a give-away: the 12 sounds included in the demo include some very nice emulations of the classic analogue sounds that shaped my formative years, and there are 1000 factory presets to play with on the $259 CD-ROM-based full program.

Merely sounding good isn't enough these days, and the *Retro AS1* comes with the sort of arpeggiation, layering, splits, multitimbrality and real-time MIDI control that you would associate with a hardware expander module. In many ways, it's probably best to just imagine that it's a screen editor for a hardware box, and you'll get on just fine. In case you're wondering about the delay between a MIDI Note On message and audio output starting, that's the kind of detailed stuff you'd find in a full review, rather than a quick mention in Apple Notes! I would say, though, that this problem would only really only occur if you wanted to play the synthesizer from an external keyboard, since a sequencer can always nudge a track back in time to compensate for sluggish responses.

The demo can be downloaded (1.7Mb each for the demo and the manual) from the Bitheadz web site, and provides enough functionality to let you assess whether the *Retro AS1* is for you. It requires MacOS 7.6.1 or higher, though MacOS 8 and up is recommended, together with a 120MHz or faster Power PC Mac, 32Mb of RAM, 40Mb of free hard disk space and a CD-ROM drive. ▶

how would you like to learn sound engineering?

this way   or   the sae way?

certificate-diploma-recording arts (hons) degree*

[26 schools worldwide · international recognition · individual practical studio time · transfer between our schools · a possible internship with disney entertainment · the best studio equipment · the largest number of working graduates · local and international job placement · sound technology course** · bachelor of recording arts (hons) degree* · multimedia producer course · bachelor of multimedia arts (hons) degree* · since 1976 · grants available to those who qualify]

london · united house north road, london n7 9dp, tel (0171) 609 2653
glasgow · unit 6/3, 8 elliot place, glasgow, g3 8ep, tel (0141) 221 3441

Main colleges for each country:
new york · from september 1998
nashville · from october 1998
paris · 45. avenue victor hugo, bat 286, 93534 aubervilliers, paris, france, cedex tel (01) 48 11 96 96
amsterdam · nieuwe kerkstraat 118, vm 1018 amsterdam, the netherlands, tel (020) 622 8790
zurich · pfingstweidstrasse 30, 8005 zurich, switzerland, tel (01) 445 2040
milano · via morimondo 19/21, 20143 milano, italy, tel (02) 8912 0540
vienna · leystrasse 43, 1200 vienna, austria, tel (01) 330 4133
stockholm · rasundavagen 45, 171 52 solna, sweden, tel (08) 730 5100
munich · hoferstrasse 3, 81737 munich, germany, tel (089) 67 51 67
sydney · 57 wentworth avenue, surry hills nsw, 2010, australia, tel (02) 9211 3711
auckland · 18 heather street, parnell, auckland, new zealand, tel (09) 373 4712
singapore · 260 sims avenue, singapore, tel (65) 741 1257
kuala lumpur · no 1 jalan usj 10/1E, usj 10, 47620 subang jaya, kuala lumpur, malaysia, tel (03) 737 0935

www.sae.edu

sae TECHNOLOGY COLLEGE

** in cooperation with Walt Disney Entertainment ®
* in exclusive cooperation with MIDDLESEX UNIVERSITY · LONDON

Apple NOTES

W www.bitheadz.com
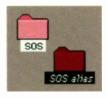
TIP OF THE MONTH: ALIASES

One of the best things to appear in Apple's System 7 was the 'alias' — that 'link' to a file that was small and could be placed anywhere on a disk in place of the real thing. They're great for avoiding navigation through lots of folders, or for providing access to lots of applications in one easy-to-use place, leaving the applications themselves, with all their support files, in neat folders. But there's at least one use for aliases which surprised some people when I showed them.

Here's the scenario: you have a set of files which are used by several different applications, and which need to be stored in a single folder, but getting to that folder from those apps requires tedious navigation, and even more tedious re-navigation when you switch between apps. One example might be some MIDI files which are used by a sequencer, and some AIFF sample files which provide some of the sounds in the sequenced song, but which are edited by an audio processing application.

The technique is simple, but remarkably effective. You do three things:
1. Create a folder and name it, using a space character in front of the name. I've called it ' SOS' for this example.
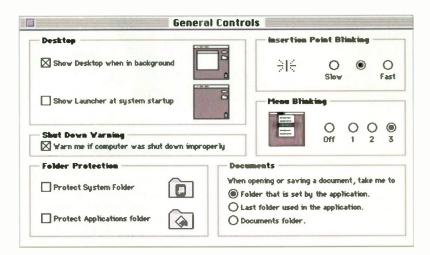
MAC HEALTH WARNING!

Sounding like a computer game, Autostart 9805 is actually a new Mac-specific 'worm' that utilises a neat feature of QuickTime 2.0+ to spread itself, virus-like, across a Power Mac's hard disks, causing lost data as it does so. After a few years of relatively quiet virus activity, this could well catch out anyone who is not prepared, and, as usual, virus-checking software needs to be up-to-date if it is to be effective. But there's a simple way to prevent infection in the first place: go to the QuickTime Settings Control Panel and turn off the two AutoPlay settings. This stops your Mac automatically running a CD-ROM when it is inserted, preventing the worm from doing anything. So go and protect your Mac NOW!

QuickTime™ Settings

AutoPlay ▼

☐ **Enable Audio CD AutoPlay**
Plays audio CDs in sequential track order automatically when inserted.

☐ **Enable CD-ROM AutoPlay**
Allows some CD ROMs to start automatically when inserted.
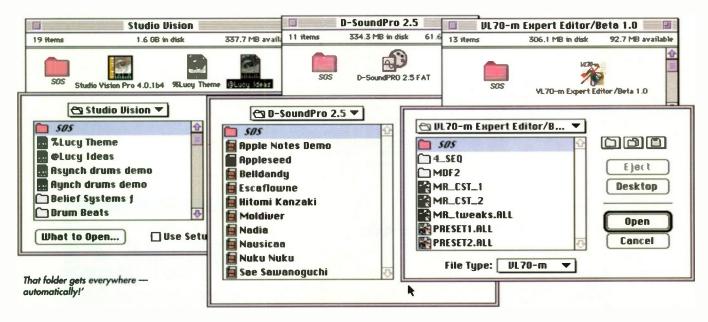
2. Make several aliases of this folder, and copy them to the folders containing each of the applications that will use files from the folder. Then rename them to remove the 'alias' bit (though you can leave it there if you want to).
3. In the Apple menu, find 'Control Panels', select 'General Controls' and make sure that the 'Documents' setting is set to the 'Folder that is set by the application' radio button (the top option).

That's all you need to do. When you next use the 'Open...' option in the File menu, you should find the ' SOS' folder alias at the top of the list, and you can get to it easily. The same applies to any application where you've copied the alias to the default folder used by the application — that's where the app first looks for files when asked to open something. Effectively, you can now access the 'SOS' folder from anywhere in a couple of mouse clicks.

General Controls

Desktop
☒ Show Desktop when in background
☐ Show Launcher at system startup

Insertion Point Blinking
○ ◉ ○
Slow   Fast

Menu Blinking
○ ○ ○ ◉
Off 1 2 3

Shut Down Warning
☒ Warn me if computer was shut down improperly

Folder Protection
☐ Protect System Folder
☐ Protect Applications folder

Documents
When opening or saving a document, take me to
◉ Folder that is set by the application.
○ Last folder used in the application.
○ Documents folder.

Studio Vision
19 items   1.6 GB in disk   337.7 MB avail
SOS   Studio Vision Pro 4.0.1b4   %Lucy Theme   @Lucy Ideas

D-SoundPro 2.5
11 items   334.3 MB in disk   61.6
SOS   D-SoundPRO 2.5 FAT

VL70-m Expert Editor/Beta 1.0
13 items   306.1 MB in disk   92.7 MB available
SOS   VL70-m Expert Editor/Beta 1.0

📁 Studio Vision ▼
📁 SOS
📄 %Lucy Theme
📄 @Lucy Ideas
📄 Asynch drums demo
📄 Aynch drums demo
📁 Belief Systems f
📁 Drum Beats

What to Open...   ☐ Use Setu

📁 D-SoundPro 2.5 ▼
📁 SOS
📄 Apple Notes Demo
📄 Appleseed
📄 Belldandy
📄 Escaflowne
📄 Hitomi Kanzaki
📄 Moldiver
📄 Nadia
📄 Nausicaa
📄 Nuku Nuku
📄 Sae Sawanoguchi

📁 VL70-m Expert Editor/B... ▼
📁 SOS
📁 4_SEQ
📁 MDF2
📄 MR_CST_1
📄 MR_CST_2
📄 MR_tweaks.ALL
📄 PRESET1.ALL
📄 PRESET2.ALL

File Type: VL70-m ▼

Eject
Desktop
Open
Cancel

That folder gets everywhere — automatically!'

PC NOTES

Just when you thought you'd got your cholesterol down, it's nothing but chips again. MARTIN WALKER wades through the new CPU options, and uninstalls some more software.

In a recent flurry of activity on the processor front, prices have been cut on all models in the Intel Pentium II range (by between 20 and 30 per cent), and new devices are also available from Intel (the Celeron 300MHz), and AMD (the 266MHz and 300MHz K6-2).

The first machines featuring Intel's Celeron 266MHz processor (released in April) are now appearing — to a very mixed reception from most people in the industry. Celeron is basically a Pentium II processor without the Level II cache which is designed to be used with a low-cost motherboard sporting the new 440EX chipset (although several manufacturers are already working on ways to fit it to other motherboards). Celeron is Intel's entry-level answer to rivals such as Cyrix and AMD because it enables manufacturers to offer home user Pentium II systems for under £700. But it's also seen by many as a reaction to poor initial sales of full Pentium II systems, since many users still seem to be finding their Pentium MMX-based PCs perfectly adequate.

Although the Celeron 266MHz chip is half the price of the full Pentium II version, most mainstream applications still run faster using Pentium MMX 233MHz and AMD K6 processors. However, because of its faster Pentium II floating-point performance, 3D games (and probably MIDI + Audio software)

will run significantly faster. In fact, tests using 3D *Winbench 98* showed that the Celeron 266MHz beat a Pentium 233MHz MMX by a healthy 26 per cent. For cheap game-based PCs, the new CPU may prove a good seller, but the problem is that because of the special motherboard, upgrade potential is poor. Until this is resolved (and with Pentium II/266MHz systems now selling for less than £1000), those looking for a new machine for music should still invest in the full monty.

THE 3RD DIMENSION
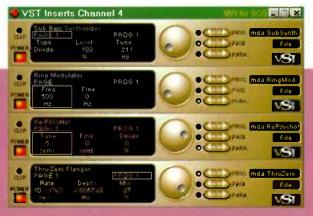
For many years, people were eager to upgrade their business systems to cope with the demands of the latest software. However, apart from more specialised applications such as computer-aided design, many business users now seem perfectly happy with the performance of their Pentium machines, and see little point in upgrading to the faster Pentium II range. After all, if you are using a word processor or a spreadsheet, most of the time the PC is waiting for user input. So what's the point of buying a faster machine?

Faced with this problem, more and more effort is being expended on selling machines at lower prices for consumer use. The fact that most people with home systems play games (even if they work at home too), means that excellent 3D games performance seems to be where most manufacturers are aiming.

THE AMD K6 2

Despite the fact that Intel are intending to drop their MMX Socket 7 processors altogether by the end of this year, the other CPU manufacturers are still pushing the boundaries of what can be achieved with this format. The ▶

Some of the excellent freeware VST plug-ins written by Paul Kellett of Maxim Digital Audio.

► AMD K6 2 is yet another Socket 7 device.

The original K6 processor range has been very successful for AMD, as for most business applications it runs faster than Pentium MMX processors at the equivalent clock rate. However, for audio processing (with lots of floating-point calculations), it runs significantly slower, as I have discussed in this column before. It's no good saying that MIDI doesn't use floating point. All music software developers now incorporate audio processing into their latest products, so floating-point performance has become important for virtually every musician.

AMD recognise the importance of 3D games, but their K6 2 processor takes a very different approach to that of Intel. Whereas the Pentium II has strong brute-force floating-point performance, 3D games tend to need a specialised set of instructions to carry out the same instructions many times over on lots of data. This process is known as SIMD (Single Instruction Multiple Data), and was used in the MMX version of the Pentium. Although great benefits were promised from MMX, only image processing really gained significantly from its integer acceleration. However, the K6 2 optimises floating-point performance, using a new part of the instruction set which is called 3DNow!

Much like MMX, the potential benefits of this new feature depend on software developers writing specific 3DNow! routines. Once again, music software is unlikely to do this, so, as always, we come back to the same recommendations. The new AMD K6 2 range provides the best value and performance for games only, and while the Cyrix 6x86MX range is ideal for the business user, it is not really suitable for musicians, due to its lacklustre floating-point performance. Currently, the best (but most expensive) CPU for music use is the Pentium II. Once initial teething troubles are over, the Celeron may prove to be a very cost-effective solution (especially for games, and possibly for music, but not for general office work).

This leaves the Pentium MMX system hanging on as an entry-level system for a few

Cleansweep Deluxe has extra Internet features over the standard version, and runs faster as well.

months more, although some retailers are already dropping these, as Pentium II prices are now so competitive. The potential choices for the musician seem therefore reduced to a list of one. Monopoly, anyone?

CLEANSWEEP DELUXE

I've been using the excellent *Cleansweep* v3.0 utility from Quarterdeck ever since I first reviewed it way back in *SOS* April '97. Its Smart Sweep utility can automatically monitor any new install on your PC, so that whether the developer provides an uninstall function or not, you can later safely remove anything from your hard drive completely (including all the impossible-to-find entries that get placed in your Registry). It can also scan previously installed applications to determine what files they use, so that thorough uninstalls are still possible. Along with comprehensive selections to clean up orphaned, duplicated or unused files, and a Registry editor and cleaner, it also has the ability to update itself automatically over the Internet.

A new *Cleansweep Deluxe* v3.0 is now available, and in addition to all these normal

functions, it also cleans up unwanted files downloaded over the Internet, including the cache (which admittedly is easy to do by hand), cookies, browser plug-ins, and ActiveX controls. It also includes *TuneUp AV*, the new Anti-Virus utility.

Cleansweep Deluxe seems significantly faster than its predecessor (especially when scanning your hard drive), with enhancements for Smart Sweep that are claimed to be more efficient and thorough. This new version is still a bit too enthusiastic at times when it suggests files that are no longer needed, but still provides Safety Sweep — a perfect safeguard in the event of anything being accidentally wiped that you later find you need. Just backup everything you delete and leave these backups on your hard drive for a few days. If an application does grumble, you can restore the appropriate file.

Although the Internet additions of *CleanSweep Deluxe* are useful, they are probably not essential if you already have *CleanSweep* (although you can upgrade to the Deluxe version for £15. Phone Quarterdeck on freephone 001 800 7212 7212, or email qservice@quarterdeck.com). This is still the most comprehensive uninstaller I have used to date, and anyone who installs software on a regular basis would find it indispensable. Most mail-order suppliers should have it at about £40, with *Cleansweep* v3 still available at about £30. By the way, although the Quarterdeck web site (www.quarterdeck.com) has details of *Cleansweep* v4.0, this is a US-specific release not available in the UK at present. **SOS**

TINY TIPS

When using audio recording software, it's not uncommon to find lots of extra files appearing on your hard drive. Of course, you can just leave them alone, but it's annoying when the clutter starts to build up, and you're not quite sure what can safely be deleted. The reason for many of these files is that although loading of audio files for playback can be done piece-by-piece as required, to first display their entire waveform on screen involves reading the entire file, which can take a considerable time. The solution is to create a waveform 'overview' — a much smaller file that allows subsequent loads to display the entire waveform quickly on screen. It's these files that can accumulate, although it is safe to delete them during hard-drive housekeeping. If the associated audio file still exists, it will simply take longer to open next time. *Wavelab* uses the suffix *.gpk (graphic peak) for such filenames; *Cool Edit Pro* uses *.pk; *Sound Forge* uses *.sfk; *Cubase* uses *.ovw (overview); *Cakewalk* uses *.wov (waveform overview); and *Logic Audio* uses *.ovm, *.ovl, and *.ovr (overview mono, left and right respectively).

World Radio History

Atari NOTES

Hardware development is still going on for Atari computers: now STe users can buy a new processor accelerator that brings faster operation and an internal hard drive to STe computers, for relatively modest cost. DEREK JOHNSON feels the need for speed...

There are many things you can do to improve the speed and performance of your ST: use an alternative OS, install an alternative desktop, buy a hard drive. You'll certainly notice a difference. But there's nothing like actually giving your computer's now-aged processor a kick up the pants, and that's exactly what the new Veloce+, from The Upgrade Shop, offers. This STe-specific accelerator has been developed especially for people who use Steinberg's *Cubase* v3 or *Cubase Score*, or Emagic's *Notator Logic* on their STs; other high-end software, such as graphics and DTP packages, will also benefit. If you're feeling frustrated at performance bottlenecks and sluggish boot-ups, Veloce+ could be the extra for you.

The new board fits inside your STe's case, mounted on top of the computer's main processor, along with an optional IDEal hard drive system. The accelerator uses a 32-bit 68020 microprocessor, clocked at 16mhz, with the excellent TOS2.06 operating system and 2Mb or 4Mb of 32-bit fast RAM fitted. The fast RAM utilises the full potential of the 68020 to give the system its real speed. The Veloce+ RAM adds one interesting side-effect: it's added to whatever you have installed in your STe. So if you have a 4Mb machine, say, adding the accelerator will increase your RAM allocation to 6Mb or 8Mb. In the rare cases where a piece of software can't see all that RAM, the original 4Mb should still be visible, and if you're running software that's incompatible with TOS2.06 you can also switch back to the original STe processor.

Pricing depends on what you buy; the Veloce+2, with 2Mb fast RAM fitted costs £149, while the Veloce+4, with 4Mb fast RAM, will set you back £189. An internal hard drive — a miniature model is used — can also be added, starting at £100 for 170Mb and going up to £160 for 720Mb. The systems can be supplied for DIY installation, or there's a fitting service costing £15, plus £7 courier collection and delivery in each direction.

If you don't have an STe and would like to take advantage of all this internally-mounted power, perhaps now is the time to upgrade your machine. It's still possible, in local 'Free Ads'-type papers, to find Ataris of all flavours for between £40 and £100 without a monitor. In fact, just as I was finishing this I found a 1Mb 520STe for exactly £40 in my area. **SOS**

A The Upgrade Shop, Unit F6, Venture House, Cross Street, Macclesfield, Cheshire SK11 7PG, UK.
T +44 (0)1625 503448.
F +44 (0)1625 503448.
E dave@tusdev.demon.co.uk

EXPANDING OPTIONS

As you may know, there's a lot of software on the Internet — much of the public domain and shareware material I discuss in this column is obtainable in this way, as well as from postal libraries. When software is distributed electronically, it's invariably been data-compressed in some way. This saves space wherever the software is being stored, and it takes less time to download, which saves you money on your phone bill, though you do need to decompress the software before you can use it. Luckily, there's a large number of tools out there for handling the different compression formats.

The majority of software I've been downloading on your behalf lately has been in ZIP format. The ideal tool for extracting files from ZIP archives is *STZip*, by Vincent Pomey; this is 'postcard ware' (just send Vincent a postcard!), so you have absolutely no excuse for not registering. You can find it in most PD libraries, or download it from the ever-reliable Ultimate TOS Software Index (http://ping4.ping.be/dipching-drulkhor/PRG-IND2.HTM).

Lately, I've also come across a couple of software archives with an LZH suffix. I do have tools to deal with this kind of archive, but they're based on a tedious command-line interface (CLI). So I had a look on the Internet for a shell program — something to provide a graphic front end to CLI software. I found it in Gregor Duchalski's *Two-in-One* shareware archive manager. *Two-in-One* provides a comfortable GEM interface for packing and unpacking all the popular archive formats, including ZIP, LZH, ARC, ARJ and ZOO. You don't really need to know what the various formats are, except inasmuch as you'll need the archiving tools themselves — *Two-in-One* merely provides the shell to software you may already have. However, registering your *Two-in-One* will provide you with the latest version of the program, a personalised key, *and* copies of all the archive utilities used by the program. Other utilities and file viewers can be integrated into *Two-in-One*, creating an easy-to-use setup capable of packing and unpacking any archive format, creating and decoding uu-encoded archives, scanning for viruses, copying, moving and deleting files, and viewing any file from any archive without having to extract the complete archive. On-line help is provided by the ubiquitous *ST-Guide* accessory.

Registering *Two-in-One* couldn't be easier. Although the software is German, it's been translated into English and is registerable via the InterActive web site, run by Joe Connor, who you may know as editor of *Atari Computing*, the sole UK-based print mag for ST users. InterActive (www.cix.co.uk/~inactive/index.html) offers a convenient way to register non-commercial software, with support for over 50 titles by 23 authors. The site is also a good place to download the latest versions of the support software. *Two-in-One* costs just £13 to register.

Joe Connor's InterActive shareware support web site makes it easy to get hold of and register a wide range of ST shareware. **SOS**

demo DOCTOR

If you'd like our resident specialist JOHN HARRIS to review your demo tape, just send it on cassette or DAT, with recording details and a photograph, to: Demo Doctor, Sound On Sound, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ.

CYBERMONKEY

Recording Venue: Home.
Recording Equipment: Pentium PCs running *Cakewalk Pro 3.0, Xgedit 95, Hammerhead, Rebirth* and *Stomper,* two Creative Labs Soundblaster AWE32 soundcards, Spirit Folio Notepad mixer, Sony Minidisc recorder, HP 7100i CD writer.

cybermonkey

Monkey seems to be the latest craze word for band names; this month I've had three monkey bands, but this is the only one to warrant a review. The band was formed last summer when they realised that most dance music at festivals was 'dull to watch' and thought that they could do better. Three vocalists front the band, for visual impact, and the music has been described as "ambient techno with the flavour of Goa Trance".

The first mix, 'Violator', begins with a lonely synthesized space whistle, *X Files* style, but a fast kick drum soon joins it and ups the pace, with breaks including a clean guitar figure which makes occasional one- or two-bar appearances. This is reverbed tastefully, and is very effective. Otherwise the track features some good programmed sounds from the Emu Classic Keys and Roland Super JV1080, and I could swear that the bass line is generated by a Yamaha soundcard. They steer clear of harsh tones and instead colour their soundscape with warm pads, octave-bouncing bass and classic dance snare and hat sounds.

Taking the ambient theme for the second track, 'Bank Job', Tin of Cyber Monkey proudly proclaims that the vocals are "untidy and muffled" and that the looped sample "leaps in and out of the mix. We love it!" The sample in question is a vocal segment from another track, and this never gets out of hand, flowing in and out of the mix with ease. The spoken vocal could have been improved with a little bass cut rather than a treble boost. It sounds as though it was delivered very close to the microphone, and has just too much of that DJ-style bass proximity effect.

Moving on to vocalist Bec's "personal favourite" mix, this kicks off on pizzicato strings with a short room reverb. It's more frantic than previous mixes, courtesy of a broken-sixteenths synth loop burbling away throughout and a house-style snare part. Bec claims to like it because she doesn't sing on it. Instead there's a gloriously deadpan vocal from programmer Skip, and this suits the the backing well.

The band like to create atmosphere, and their attention to detail in the choice of sounds, especially on the more sparse 'Sleepy Song', is excellent. Overall the CD has a fat, warm sound. I think I'll keep this one for my collection and look forward to playing it again! ❑

ABRAHAMS MERKIN

Recording Venue: Home.
Recording Equipment: Steinberg *Cubase* sequencer running on an Atari STe, Soundcraft Spirit Folio mixer, Akai 4000d open-reel stereo recorder, Electroharmonix Small Stone phaser pedal.

The intriguing title of this demo is *Swarfega Dreams*, which leads one to assume that its author has had dealings with oily products in the past (and yet he's still not managed to get a record deal so far!). The opening does feature a nasty earth loop hum, which I presume is on the sampled vocal and could have been cleaned off — not with Swarfega, but with a noise gate. As Abrahams doesn't include one on the equipment list, some low-frequency roll-off would have reduced the problem without affecting the vocal sound too much. Experimenting with the LF EQ or bass roll-off switch on the Spirit Folio desk should give the desired result. The underground sound of the bass end on the first mix is very low and is right on the verge of distortion — but in a nice way. Unfortunately, there's not much action in the presence area of the mix in some sections, and even a hefty boost of 8dB at 12kHz isn't going to remedy this problem without exposing the digital crunch. Going for more continuity in the overall sound wouldn't be a bad idea, as in some areas there's enough treble while in others there's very little.

The second track shows that this can be done with the gear available and almost goes too far in the other direction. The bass sound on the first track, for example, is beautifully warm and full, and would have suited the second track as a perfect foil to the more percussive upper-mids in the mix.

As it is, the twanginess of the more 'open-filter' bass synth competes with the other sounds on the track a bit too much for my taste.

Some of the 'locked' sections of sound, whether done on the Akai sampler or the sampling delay, are effective heighteners of tension as they build up into a wail of sound over the dub bass line and heavily echoed percussion track. And the very end of the track sounded very much like a tape echo dying away, which I suspect was the intention, given Abrahams' liking for Electroharmonix pedals! ❑

Recording Venue: University College, Warrington.
Recording Equipment: Soundcraft DC2020 mixer, Tascam DA88 and 38 digital 8-track recorders, Steinberg *Cubase Audio* sequencer, Digidesign Pro Tools Project (8 tracks), Drawmer 1960 tube compressor, Drawmer DS201 and DL241 gates, Digitech Studio Quad effects, Aphex C2 Aural Exciter, Lexicon LXP15 effects, Yamaha SPX990 effects, TC Electronic Finalizer mastering processor, AKG and Shure mics.

This CD is a limited-edition compilation from the students of University College, Warrington. Part of their final year in Media and Business Management involves producing a CD featuring some of the North West's hottest unsigned talent. This is the fifth CD release from the college, mixed and produced on campus in the reasonably equipped college studio.

A synthesized string arrangement kicks off the CD (I expect to be hearing more stuff like this in the future, given the success of The Verve). The strings actually sound well arranged and recorded, with plenty of presence in the sound, so it comes as a bit of a shock when the rock-orientated track, by Halcyondaze, really kicks in. It's very heavily compressed, and has rather had the life sucked out of it with what sounds like a limited bandwidth, until the vocal starts. This and the sizzly hi-hats manage to put back into the track what was missing in the high frequencies.

The sound improves for the next track, by Dogtomas. This is less compressed, with a more open drum sound, and some of the harder edge in the mid frequencies has been retained. If anything, the hi-hats are over-equalised and tend to dominate the mix, so it comes as a welcome break when the drummer stops playing them for a few bars. Bringing the vocals and guitars up in the mix would have given it more energy — as demonstrated on Giro's song 'I Never Learn', mixed by Scott Atkinson, James Blake, Curtis Darroux and Aaron Obita. This recording brings out the garage band element of this Liverpool combo, although once again I suspect slightly too liberal use of an exciter somewhere along the line, over-emphasising those hi-hats.

Steve Seager's track 'Cage' comes out best on this worthwhile compilation. It's a strange mix of northern grunge and drum machine, with a haunting chorus melody line. The reverb is not used sparingly on the kit sound, but its metallic tones add an industrial flavour to the mix that works well on the chorus.

It's a fine idea to organise, engineer, record, mix and market a compilation CD as part of a college project, and if it helps a few bands along the way that's great too. Generally, the main faults in the mixing appear to be from over-use of the exciter and compressor and a low vocal level, but mixing by committee is never easy! Otherwise, it's good stuff. ❏

WIN AMPEX TAPES

Ampex Professional Media
by QUANTEGY

As an incentive to send in your best demos, Quantegy, makers of Ampex tapes, are kindly providing prizes for the best demo tape submitted each month. The lucky winner has a choice of: 20 x Ampex 472 90-minute chrome audio cassettes, 10 x Ampex 467 DAT 90-minute cassettes, 3 x 456 or 2 x 499 half-inch open-reel tapes, 5 x 456, 457 or 499 quarter-inch open-reel tapes, or 5 x 489 40-minute SVHS ADAT cassettes.
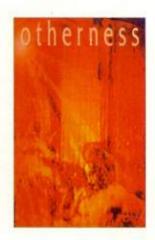
Recording Venue: Home.
Recording Equipment: Atari ST running Steinberg *Cubase* sequencer, Roland VS880XP digital recorder, Roland S760 sampler, DDA Interface desk, Tascam DAT machine.

Keith Aubrey and Linda Conboy are going for the contemporary mix of beats, sliding bass, guitar and synthesized sound collages which is currently popular. We do get quite a few demos of this kind sent in, and comparisons with successful bands in the 'moody groove' vein will be inevitable. On a more positive note, this one stands head and shoulders above most of the others, for two reasons. First, they're not just reliant on the moody numbers, but have other strings to their collective bow. From the guitar-based 'How Do You Get Through Your Day?', for example, it's plain that singer Linda is capable of more than just sultry soul and can handle an upbeat slice of pop. This reluctance to slavishly emulate already successful bands should help them in their search for a record deal.
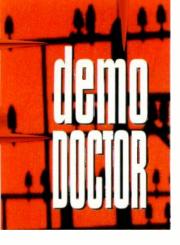
The draw of the east seems quite strong for many *SOS* readers, if the demos we receive are anything to go by, and on this tape's fourth song Otherness have succumbed to it. Some of the synthesized sounds are squeezed into that region around 1kHz where many middle eastern instruments have their main strength. This could cause a possible conflict with the vocals, but is easily avoided by leaving such sounds out of the mix when the singing occurs. An eastern melody is handled by a fatter string sound, which also avoids a conflict in the soundscape. This is a fairly typical example of the way in which the band handle mixes.
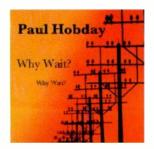
On the final song, 'Epitaph', the band are able to give the impression that a lot is going on, with a wall of sound that swirls in over the beat when the vocal line drops, only to fade down when the vocal line returns. This could have been achieved using a number of methods — simple use of attack time on samples, manual fades to tape (more time consuming), MIDI-controlled volume changes, or 'ducking' using a compressor or gate.

All the songs are good and achieve the mood that the band are after. They should do well if they tread their own musical path. ❏ ▶

demo DOCTOR

Paul Hobday
Why Wait?
Why Wait?

PAUL HOBDAY's instrumental album is mainly guitar orientated but features some nifty programming on the backings. Anyone who forks out for a PRS guitar takes their playing seriously, and Paul's ability is never in question on this album, with some tasty soloing and occasional self-indulgence. He tends to go for the heavily processed guitar sound, but it suits his style of playing admirably. Some over-hasty level changes occur from time to time — for example, the lift of the guitar solo track in the middle of the second composition, 'Flying'. What intrigued me most was how Paul would tackle Joe Zawinul's track 'Birdland', so I took the liberty of speeding on to that piece and was pleasantly surprised by the tasteful use of harmonised guitar, both clean and overdriven. The keyboards are well played, and the bass is unexpectedly decent on synthesizer, with the higher phrases given to the guitar.

JAGUAR MOSES: *White Religion* is the title of this CD, written and recorded by Will Hogan. Drums are programmed but it sounds as though

White Religion
Jaguar Moses

everything else on this rocky album has been played by Will himself. He's a good guitarist, at his best on the heavier pieces such as 'The Arrival', where he's able to fill in with some squealing runs and harmonics. It's obvious that he's a little uncertain about the vocals because of the amount of echo effect added to them, plus the double-tracking and the low level in the mix. The better vocal takes are on the thrashier numbers, although the singing and performance is nice on the slow 'Remember the Baby's in Bed'.

DA CAPO: A short atmospheric intro leads into the thumping four-on-the-floor of the opening mix. The kick has been heavily EQ'd for bass and is a bit too much for the track — this is the sort of weight that a big playback rig would allow you to hear anyway. As an instrumental it's fairly sparse, with little use of modulated effects, and this makes the track sound under-produced. Echo is used on the drums, and occasionally the vocals and a backwards vocal lend a touch of humour to the proceedings, as well as an interesting sound texture. The more eastern second mix features some nice synthesized sounds, in an attempt to recreate middle-eastern instruments. The strings lack presence but have a nice lower-mid warmth, and once again the bass end of the mix sounds too heavy — possibly a monitoring problem?

GUY LANCASTER: *Castle in the Sky* sounds like a dubious title for a tape to me, unless you're a closet Rick Wakeman or Moody Blues fan, which I suspect Guy is. However, you can't knock the playing, which is excellent, especially the keyboard and bass guitar work, featuring plenty of sounds you'll know and love if you're a progressive rock fan. Some of the pieces are heavily programmed, but there's always

Guy Lancaster
Castle In The Sky

enough real playing to give rhythmic feel to the songs. I particularly like the bass guitar sound on some tracks, which have the classic upper-mid twang and fat bottom end found on many prog-rock classics (some of which, I have to confess, may be found in my vinyl collection). I think the mixes have been given extra punch with compression, and there's a nice presence to the mix — both probably added in post production. The album consequently has a consistent production sound which many other demos we receive fail to achieve. Occasionally delving into the more modern ambient style of dance production brings some of the tracks more up to date, but Guy's strength is definitely in the melodic rock vein.

MANTIS call their demo *Acidhead* and set out their musical stall in unambiguous fashion. The first track, 'Slimy Mackerel', is lo-fi techno rock employing the recognisable sound of a tightly tuned jungle snare. I couldn't work out what instrument was taking the heavily distorted riff — it could have been guitar, bass guitar or synth. This element is effective on the first track but even better on the second, 'Beats on Toast', a powerful production of breakbeats and riffing, with some nice triggered noise-gate vocal chording going on too. All the sounds sit together well in the mix, when the temptation might easily have been to give any one of these parts a bigger, more dominant role. Good ideas, cool sounds and silly titles — a winning combination!

SONICA: Simon Cantrill is the man behind the name Sonica. His

MANTIS
ACIDHEAD

compositions are instrumentals with the occasional sampled 'ooh', and the instruments are balanced well, yet the production sounds a bit sterile. The triggered keyboard parts, especially on the first mix, are lacking in effects and therefore sound 'proud' of the guitar parts, which are heavily echoed and modulated. In the places where the keyboards are slow-attack, long-release string or muted brass sounds this is fine, and there's a nice sonic blend. With sharper, fast-attack timbres, such as piano, the effect is not so good, and where the piano is exposed it really does sounds a little dry and up-front. On the positive side, there are some nice guitar textures, very reminiscent of Fripp and Eno, which I enjoyed tremendously.

Sonica

LEON states that his influences include Kraftwerk and Pink Floyd, but I find that his music veers towards the former, being reminiscent of their more commercial work. He's certainly captured the sound of the '80s in this demo — highly compressed, punchy kick and snare, sixteenth-note hi-hats and electronic tom breaks. The mix sounds fat and analogue on the first track, with warm strings in a looped pattern carrying the main melody. In contrast, the second instrumental has a harder edge, sounding close to early techno but still managing to maintain the big sound, courtesy of the bass, a synthesized slap type with plenty of bottom end. The mixes themselves are very good and some of the riffs have that touch of humour which Kraftwerk managed to inject. If the arrangements were a little shorter, or sometimes included a vocal, it wouldn't detract from the general '80s feel. ❏

Leon.

K5000
ADDITIVE POWER

World Radio History

brief encounters

CAIG LABORATORIES DEOXIT AND PROGOLD CONTACT ENHANCEMENT TREATMENTS

As studio systems become more complex, analogue signal degradation caused by poor signal connections becomes a serious problem, especially in vulnerable areas such as patchbays. Conventional contact cleaning sprays often make the problem worse, because many use a sticky, greasy formulation which attracts and traps dust. A better solution is to use one of the more advanced contact enhancement treatments such as Tweek, Stabilant 22 or Deoxit, the last of which is under test here.

Deoxit claims to improve conductivity by cleaning away thin films of surface oxide and by preventing further oxidisation. It is not sticky or oily, and unlike some of the very expensive contact enhancers, it is cost-effective to apply by spray, though it also comes in applicator pens and as pre-soaked wipes.

For general-purpose use, Deoxit comes as a 5% solution in what seems to be an alcohol base. However, it is possible to buy a metered dispenser of 100% Deoxit as well as full-strength wipes, syringe dispensers and pen applicators. For the purposes of this review, I used the Deoxit D5 Mini Spray, which contains 20ml of 5% Deoxit powered by a CFC-free propellant. I also tried some of the wipes, which are very convenient for treating jack plugs and similar exposed connectors.

Intermittent patchbay connections are a fact of life, especially if you use budget quarter-inch jack patchbays, but wiping or spraying the jack plug with Deoxit prior to insertion cures most problems instantly. Dirty normalising contacts are a bit trickier to treat, as you need to get behind the patchbay and spray directly into the contacts, but this again brings about a rapid cure that seems to last for a long time before treatment is needed again. Other areas that benefit from Deoxit cleaning are guitar jack sockets and the phono connections on the back of DAT machines and soundcards. It may even be beneficial to treat the earth pins of your mains plugs as a low-impedance ground path is vital to effective screening.

For gold-plated connectors, Caig Laboratories have developed ProGold, which is a little more 'tweaky' than Deoxit. The benefits are similar, but in addition, ProGold claims to be able to penetrate the microscopic pores found in thin gold plating, allowing it to bond to the metal beneath, and preventing corrosion building up between the base metal and the plating. Tarnishing is prevented, and as with Deoxit, there's a full range of applicator types and a choice of 5% or full-strength solutions. Again, I tested it with a 20ml Mini Spray and some wipes. The immediate benefits are similar to

those of Deoxit, and in the case of badly tarnished connectors, the difference in audio quality can be quite noticeable. ProGold could be usefully applied to gold jack sockets, gold phonos and circuit board edge connectors, although it is important not to spray any kind of cleaner in the vicinity of a capacitor mic capsule due to the extremely high impedances involved.

Given the effectiveness of these cleaners, Deoxit in particular should be a part of every studio owner's survival kit. Inexpensive sampler kits of both ProGold and Deoxit are available containing both 20ml sprays and wipes. *Paul White*

£ Deoxit starts at £6.46 for a 16ml spray of 5% solution, while ProGold starts at £10.81 for 16ml of 5% solution.
A Probus Electronics Ltd, Findon, Southill Lane, Eastcote, Pinner, Middlesex HA5 2EQ, UK.
T +44 (0)181 866 7272.
F +44 (0)181 866 2999.

AUDIO TECHNICA ATM87R CONDENSER BOUNDARY MICROPHONE

Boundary Effect microphones have been around for some time now, and the standard PZMs (Pressure Zone Microphones) use piezo technology. The ATM87R, however, is a full-blown condenser mic, giving it the potential to display a much smoother top end than conventional piezo designs. The ATM87R has a hemi-cardioid (half-space cardioid) polar pattern and, as the name implies, has been designed to cope with high SPLs, making it suitable for very dynamic signals. The ATM87R runs off 48V phantom power, and has a two-position switch for flat response or low-frequency rolloff to control ambient noise, recessed underneath the mic to avoid accidental operation. A 25ft cable is supplied, with a miniature TA3F connector on the microphone end and the usual standard-sized balanced XLR male type on the other.

As with any boundary/pressure zone microphone, the symmetry and area of the mounting surface directly affects the sensitivity of the microphone at low frequencies. These microphones make use of reflected sound and

need to be mounted on a flat surface like a table top or wall, with the front of the mic facing the sound source along the longer dimension of the mounting surface. Sensibly, the bottom surface of the ATM87R is made of a clingy rubber material which helps the mic to stay put and also dampens unwanted noises. Overall, the construction of the microphone is excellent, though the cable supplied, with its miniature connector, is a little flimsy.

I used the ATM87R on every session I did over a couple of weeks in the studio. On piano, used subtly, it gave a lovely warmth and sense of space without removing the stereo image of the other mics, and on drums it gave just enough of the room sound without the splashiness that can come with other PZMs. I even put the ATM87R directly in front of the bass drum on one session, getting excellent results. On flute and saxophone, the ATM87R gave such a tight sound (in a fairly dead area) that it was hard to believe a boundary microphone was being used. In a small home

studio, I would think that the ATM87R would be most suited to acoustic guitars and drums — though anyone seriously thinking about making their own drum samples would find this microphone ideal for powerful 'roomy' sounds on rock tracks, or nice tight ambience for jazz and soul samples. When recording electric guitar through an amp, the ATM87R would always be useful as a room microphone, as it responds well to high SPLs and retains a nice, smooth top end.

In all, I was hugely impressed with this mic. Boundary/PZMs have always been regarded as slightly limited, specialist mics — but this one is different. If you are in the market for condenser mics, have a listen to these before making a decision — you'll be glad you did! *John Verity*

£ £199.95 including VAT, XLR cable and soft case.
A Audio Technica, Technica House, Royal London Industrial Estate, Old Lane, Leeds LS11 8AG, UK.
T +44 (0)113 277 1441.
F +44 (0)113 270 4836.
E sales@audio-technica.co.uk ▶

brief encounters

As CD-writers become more popular, so the need to label one-off CDs more professionally grows, and a number of inexpensive labelling systems have emerged to meet that need. One such is Neato, a very simple manual system that enables self-adhesive, ink jet/laser-printable labels to be centred and stuck to the back of your disc accurately and easily.

Supplied as a boxed set, the basic Neato kit provides an alignment jig, 28 assorted plain colour labels (two per A4 sheet), a couple of inlay cards, one sheet of laser-printable clear labels and a CD-ROM containing printing templates and useful artwork. The software is compatible with both Mac and PC platforms, and templates are supplied in *Quark Xpress*, Adobe *Illustrator* and *Photoshop* formats, though you can make your own with any graphics program of your choice, and if you test the printing alignment on photocopies of the pukka labels, you can perfect your work without wasting any labels. Incidentally, replacement labels work out at around 10p per disc, which is pretty cost-effective. Each sheet includes two rectangular labels that can be stuck on the outside of jewel boxes if you don't want to make a complete inlay card.

The mechanical part of the system is ridiculously simple — there's a circular plastic base with a hole in it, and a hand-held centring jig onto

which the CD being labelled is placed. After printing, the label is peeled off the protective backing and placed face up on the base. Next, the conical end of the centring jig, onto which the disc has been threaded, is placed in the centre of the label. You then simply push down. Generally, this results in a perfectly fixed, properly centred label with no air bubbles, but I have found that the labels tend to curl pretty badly when they're peeled off the backing. This doesn't usually affect the labelling process, but in severe cases, it can affect the centring slightly. When the labels are properly applied, they're claimed to be sufficiently well balanced to be used with high-speed CD-ROM drives

Neato's system is very cheap, it's easy to use and the end result looks very professional. Replacement stationery is also sensibly priced, but the company do need to look at ways of making the labels curl less, and it would also be helpful to include templates for other common programs, such as Claris *Works* or Microsoft *Office*. Despite the minor niggles, though, this is a nice little system that makes your one-off CDs look very professional indeed. *Paul White*

> **£** Basic kit (contains applicator, design software, 10 white and 2 coloured labels, 1 laser label, 2 jewel case inserts) £35.19 including VAT. Various kits are available; contact Neato for details.
> **A** Neato, College House, 17 King Edwards Road, Ruislip, Middlesex HA4 7AE, UK.
> **T** +44 (0)990 561 571.
> **F** +44 (0)1895 639955.

Keyfax's *Twiddly Bits* soundware has a very respectable track record in the market of producing MIDI files for general use by programmers and writers alike. You are supplied with a 3.5-inch floppy disk containing

bundles of short phrases, drum patterns, fills, and chord progressions ready to be loaded into your MIDI sequencer, either as a springboard for your own song ideas or so that you can sneakily pass off the competently performed series of solos and keyboard licks as your own playing. Volume Nine in the series, *Jazz*, takes four main instrumental styles associated with jazz performance (Drums, Bass, Electric and Acoustic Pianos) and supplies you with tons of short MIDI files containing versatile phrases, loops and grooves.

As the wonderful demo song file shows, this release covers an astonishingly wide range of jazz tempos and styles, from cheesy ballroom to cool latin and frantic acid jazz. It is also clear that the programmers and performers used on this release have an enviable understanding of jazz expression, rhythm and harmony, which they have translated smoothly into the often stilted and musically tepid domain of the General MIDI module.

In most cases, the phrases are played in C, leaving you with plenty of scope to transpose the parts to fit your track, and most usually work out being between two and four bars long. The phrases are naturally designed to be used in connection with a GM module or soundcard, and

in this context they work fine; however, it's when you use them with more exciting sound sources, such as a sampler or dedicated synth module, that their real musical potential is unlocked. The bass lines are of particular note in this respect, and are all tightly performed (via a real MIDI bass guitar) with just the right amounts of pitch-bend, modulation and expression. Fused with a great acoustic bass multisample, they would probably convince even jazz purists that they were listening to the real thing.

The laid-back, almost sloppy electric pianos are equally impressive, and are played with loads of thick chordal movement and sensitivity, and both the electric and acoustic piano solo phrases simply ooze attitude. I can imagine these latter files in particular will be real life-saver for the less technically able player looking for an inspirational bolt of realism in, say, the middle eight of a track.

The drum patterns, though very convincing, occasionally suffer from sounding as if they have been quantised a little too tightly. That aside, the selection of drum performances is authentic and, of course, offers the user the chance to tweak and add any elements that you feel might be missing on the original files.

Finally, even the sleeve notes are faultless; each file is listed with appropriate harmonic and stylistic information, as well as a few tips on how to place the parts within a track and ideas on possible sound types. With such an immaculately performed and logically put together soundware product, anyone looking for a touch of jazz inspiration can't possibly lose. *Paul Farrer*

> **£** £24.95 including VAT.
> **A** Keyfax Software, PO Box 4408, Henley-On-Thames, Oxon, RG9 1FS, UK.
> **T** +44 (0)1491 413938.
> **F** +44 (0)1491 413938.
> **E** mail@keyfax.com
> **W** www.keyfax.com

HOT NEW LOOPS & SAMPLES FOR SUMMER

The big news in sampling this summer – tons of HOT new releases from East-West, USB and Zero-G – check them out in one of our 75 jukeboxes at a store near YOU!!

▼ East West (USA)

BLACK BUTTA
(9/10 REVIEW)
Produced by Madjef Taylor for East West.

MadJef Taylor's album credits include artists like Janet Jackson, Michael Jackson, Karyn White and many of the albums produced by Jimmy Jam and Terry Lewis.

This has to be one of the phattest collections of dope beatz and hip-hop grooves available anywhere. Madjef has compiled a fresh variety of beats, loops, scratches, bass and lead lines, live drums and fills, kicks, snares, and guitar samples in construction kits with MIDI-files. An absolute must for any Hip-Hop or R&B producer. His unique drum programming will put a groove in your tracks that will give you the edge you need to make great records. AUDIO CD plus MIDI-files: £59.95

DRUM N' BASS CONSTRUCTION KITS
Produced by James Bernard for East☆West. This BEST SELLING 2 CD SET has the most amazing collection of Drum 'n' Bass grooves and breakdowns all in construction kit format. It contains over two hours of the latest drum and bass grooves by NYC producer and sound designer James Bernard. First you get the complete loop, followed by the drum part, sub-bass part, synth pads and individual drum hits - everything is broken down so you can use as much, or as little of each loop as you want. This collection is huge value for money – you get two CDs for the price of one! 8/10 Review – KEYBOARD, USA. 2 CD-AUDIO Set: £59.95

HYPNOTICA
Produced by Perry Geyer and Greg Hawkes for East☆West. "Fat phase-shifted and ring-modulated filter sweeps, eerie vibrato tones bouncing through a delay line, Theremin glissandi, chuffing machine clanks, crispy white-noise explosions, distant klaxons buzzing, whippy burbles, cavernous whispers - it's all here. The production values on Hypnotica are almost too high. Many of the samples consist of two or three layers of sound, almost like mini-production pieces of the trance/ambient variety. Reverb and chorusing add richness and space to a mix that glistens with high end. Truth be told Dr. Who never sounded this good. 8/10" (Keyboard USA)

New from the creators of "Technophobia!" and "Tekno/Industrial" – HYPNOTICA is loaded with fresh sounding cutting edge textures, trance and ambient loops, FX, Mind blowing Sweeps, Galactic Noises, Hypnotic sequences and magical sounds for your next rave. HYPNOTICA features complete soundscapes, ambient atmospheres, electronic sample and hold, reverse FX, noise loops, underwater textures, electronic ethnic voices, ring modulation, liquid audio, lazer sweeps, takeoffs, pans, white noise FX, low frequency sequences, oscillators, android heartbeats, drones, screams, synth bleeps, intergalactic textures etc. Orb meets Future Sound Of London - a magical mystery tour! CD-AUDIO: £59.95

SMOOV GROOVES
Produced by Michael Bland & Sonny Thompson for EastWest

"This 62-track CD doesn't bother with any demos or clumsy spoken introductions: it just kicks off with the first of the loops-and what a great way to start. Slow and meaningfully funky vibes seem to pour out of every note and, despite being more laidback than an arthritic tortoise on Mogadon, the timings, as you might expect, are absolutely spot-on.

The loops range for 60 to 108bpm, and vary from sort of camp '70s cop show feel right up to the familiar spaced-out jamiroqual-esque MTV style of the '90s. The recording quality is as good as the excellent playing. With these guys as his rhythm section Prince can justifiably sing "My name is Prince, and I am Fon-Keh", and with this CD in your collection you can now say the same thing. FIVE STARS our of five ☆☆☆☆☆". (Sound on Sound, UK)

From the rhythm section of PRINCE and the 'New Power Generation' - Michael Bland & Sonny Thompson. SMOOV GROOVES is the sequel to the critically acclaimed Funky Ass Loops CD and contains all the loops you need to lay down everything from slow grooves to 70's funk (the tempos range from 60 -108 BPM). First you get a full mix of each loop, followed by the drums, bass, guitar, percussion, keys etc. SMOOV GROOVES is without doubt the finest collection of smooth laidback grooves & 70's funk available anywhere for any price!!! - and it comes to you from one of the world's funkiest rhythm sections - Sonny T. & Michael B. Audio CD: £59.95. CD-ROM (Akai or Roland): £99.00

▼ East West (USA)

ELECTRONICA
Produced by Perry Geyer and Greg Hawkes for East West. "With a range of tempos from 68bpm to a pounding 214 bpm, the spectrum of applications for these samples is reassuringly broad, and as a mixed mode CD, Electronica not only gives you audio samples - both as loops and individual hits - but also all the loops as MIDI files, allowing you to load the data into your sequencer and adapt or adjust the loops for yourself. The loops are generally excellent, and varied enough, both rythmically and in terms of effects and production, to indicate that some serious time and effort has gone into this disc. Verdict – There's a broad range of applications for these well produced samples. The MIDI files are a big bonus as well. 9/10" - Future Music (UK). In this collection you'll find electro industrial loops & samples, many with MIDI-files (enabling you to recreate & customize these loops). There are two types of loops on ELECTRONICA, complete loops without MIDI-files that were generated 'live' in the studio using a combination of sequencing, synthesis & FX generation, and those with MIDI-files. The first group are complete, without any components; the MIDI-files loops start with the complete loop followed by each individual loop component. There are also many individual sounds at the end. Why do you need the MIDI-files and individual samples? Suppose you're working on a track, and want to combine a few different loops at a particular tempo. You can now do it using the loop sequence data from the MIDI-files to adjust the tempo of each loop in the sequencer - without altering the pitch of the individual samples! Or, you can replace individual samples or parts of the sequences to create new 'custom' loops featuring your own samples. We know some of this is possible with programs such as ReCycle , but you have much more control with the sequence data & loop components. If you don't need this feature you simply use the main loop. The musical content of ELECTRONICA is cutting edge electro industrial for the 90's and beyond. CD-AUDIO & MIDIFILES: £59.95

THE ULTIMATE STRING COLLECTION
(East☆West). The Ultimate String Collection is the new standard for string libraries. The collection is perfectly recorded and includes many unique features that put it in a class of its own. Apart from its superior sound quality, The collection features up to 34 violins (for a rich warm sound), 8 violas, 8 celli, and 6 basses. Most of the sections were recorded with double bow action - up and down strokes. This feature provides unparrelled realism for fast sequences. Every section has been recorded with up to 3 layers. Two different layers per tone are provided for every section, pizzicato, marcato, plus the very best tremolo. The Strings have been recorded completely dry to enable users to choose their own ambience. Akai CD-ROM: £199.95. Roland CD-ROM: £199.95

THE ULTIMATE PIANO COLLECTION
(East☆West). REVIEWS: "The Ultimate Piano Collection is, in my experience, is as close to the real thing as technology will currently allow" Sound on Sound (UK). KEYBOARD agreed, awarding the collection 5 out of 5. This collection contains four of the best Pianos ever built: Steinway D, Steinway C, Fazioli F228, and Bösendorfer 225. Every key has been sampled in various attacks, with and without sustain pedal. There are 13 different kinds of piano sounds each with its own characteristics. These pianos start sounding real good with 24 Mb or more of RAM (16Mb minimum is required). Loads up to 64Mb are provided - the current industry standard. CD-ROM (Akai or Roland): £199.95

SCORING TOOLS
(East☆West). Immaculately recorded with a sixty piece world class symphony orchestra. SCORING TOOLS provides you with ORIGINAL orchestral phrases, beginnings and endings, in 3 different keys, 2 different tempos and 7 different styles of music including Adventure, Fanfare, Romantic, Magical, Mysterious, Suspense and Western. All you have to do to create finished tracks is add your own melodies. There is nothing else available, other than hiring a symphony orchestra and writing and recording the arrangements yourself that enables you to achieve such realisic results - this is the most powerful and inexpensive tool available to create ORIGINAL orchestral soundtracks. 9/10 REVIEW – KEYBOARD (USA). Akai CD-ROM: £199.95. Roland CD-ROM: £199.95

JUNGLE FRENZY VOLUME 2
Hot on the heels of the best selling Jungle Frenzy... The ultimate resource for drum & bass musicians. Jungle Frenzy 2 picks up where Volume 1 left off.

Tons of new breaks, transformations and cuts, basses, pads, fx and vocals, infact - everything you'll need to make your mark on the Drum & Bass world. CD-Rom (Audio, WAV & AIFF) £19.95

▼ Zero-G (England)

CHEMICAL BEATS
(Zero-G)
The Problem: Drum Loops are easy to use. Load them from CD or CD-Rom, adjust the tuning if necessary and away you go. Unfortunately loops are difficult to customise to your needs, so you never seem to have enough of them to find the perfect groove or break. Separate drum samples give you flexibility. You can change sounds, change rhythms and add FX and EQ to individual parts, but inspiration can fade away while you look for the right samples and program the right feel. Wouldn't it be great to have the convenience of loops with the flexibility of separate sounds?

The Solution: With the Beat Generator System™ you can have the best of both worlds: Imagine fitting 350 unique Stereo killer drum grooves into 16Mb of sampler memory. Hold down just two keys on a standard 5 octave keyboard to hear any combination. Think about being able to separately drop out the bass or snare drum on the fly, without missing a beat, or substituting a different sound to play the same rhythm. Consider the possibilities of adding individual bass and snare drums, changing the feel by offsetting or re-triggering the separate loops which make up the groove, quickly and intuitively. Now imagine all of this power multiplied by over twenty and you start to see the depth of the system.

Chemical Beats: These are loops with masses of attitude. Huge drum sounds and FX crunched through advanced signal processors. Massive sounds to kick start your imagination. Years of playing, programming and engineering expertise at your fingertips, but so flexible that you may never run out of new rhythms and sounds to try! To complete this awesome package, there are extra mad percussion loops, thunderous basses, radio loops, cutting edge synth & guitar FX, plus a vast collection of drum & cymbal bits. If you're into BIG beats, this is the CD-Rom you've been waiting for! Double CD-Rom for Akai S3000 series £119.00. Audio CD (Selection of loops and sounds) £59.95 (Due for Release in June/July '98)

CUCKOOLAND "ASYLUM"
(Zero-G). The third and final chapter from the loonies who bought us the classics 'Unhinged' and 'Ghost In The Machine'. Asylum is the most insane offering yet.

A wild collection of strange loops, crazed voices, twisted noise and just plain madness. Asylum is the most comprehensive of them all! In 'Ward A' you will find BPM grouped loops that will have the same effect on you as a month of electric shock therapy - in other words you will discover large selections of noises that make programming your own grooves and atmospheres in a Cuckooland style a piece of cake. Asylum is un-nerving where Unhinged was stange and Ghost In The Machine was scary. Why not visit the Cuckooland Asylum... your music will never be the same again. Audio CD: £59.95. (Due for Release in June/July '98)

PLANET OF THE BREAKS
(Zero-G). It has become increasingly difficult to unearth both rare and unused drum breaks - in fact, almost a mission impossible. Planet Of The Breaks introduces a new generation of 'Raw-funkin-dirtyass-boogiebeats' to sample. Original loops that mix old school style and sound with the sonic science of the new school. Drums recorded through vintage analogue and valve gear, then digitised, sliced, diced, tweaked and freaked. This ain't no ordinary sample set by some famous drummer. This shit is dope! Each break has been individually created with mad love using a wide variety of production, miking, processing, recording and sampling techniques. Practically every make of drum kit out there has been played and recorded at twenty different studios in both London and Los Angeles. Variety of sound is a key element in any CD and a huge effort has been made to deliver lots of raw breaks for your own sonic interpretations and mutilations. Hell, Zero-G even cut to vinyl and mastered off of that! Planet Of The Breaks delivers! OUT NOW!! Audio CD: £59.95.

▼ Ilio (USA)

TRANCE FUSION
How do you make that great energy-building, heart-pounding climb from one killer groove to the next? Or add drama and intensity to keep your tracks from just sitting there? TranceFusion is 100's of noise sweeps, spectral climbs, glides, fills & morphs to carry your remix from one groove to the next, or simply to add interest to a steady pulse. Also perrrcussion sweeps (32nd notes), swell-stabs, and big hits to spice up your transitions, & rave and trance style arpeggios, pads & other madness. The sounds were created using a war chest of the hottest vintage & analog gear, from classic VCOs to physical modeling & every wavetable-vectored-linear-additive-frequency-transformed-pulse-code-modulated step in between. Truly a one-of-a-kind, TranceFusion will force its way into your head and never let go! CD-ROM: £149 (Akai/Emu/Kurzweil, Roland, SampleCell). AudioCD: £59.95

▼ U.S.B (France)

DOPE HEADZ
This CD features hundreds of acoustic drum loops played with a strong Jungle / Trip Hop attitude. The drumming skills displayed here are for real but they defy imagination : jungle-style 160 bpm loops with a live feel, truer than real Dub loops, etc. You have to hear it to believe it! USB's goal was to provide you with loops which are very natural-sounding, as well as loops with weird and distinctive treatments. Running these loops along sequences will bring a massive dose of life in your music. As an additional help, drum hits were recorded separately during the same sessions, so that they could be included as samples in the bonus section. Certainly one of the most distinctive an addictive drum library on the market. Audio CD: £59.95. Akai/Emu/ASR CD-ROM: £149.00

KITCH 70's
Kitch 70's is the biggest collection of 70's Drumloops and Samples recorded with the best vintage equipement. Thanks to digital surgery, these loops and samples ooze authenticity.

Kitch 70's is available in Akai CD-ROM format with MIDIfiles as well as audio CD. Each loop has a specific tempo and is precisely organized : a huge task to create a unique CD faithful to the sound of the seventies. The emphasis has been put on the extreme audio quality of this CD, using top of the range equipment and record players. A real jewel for Groove, House & jungle. This is for all of you dance producers! Audio CD: £59.95, Akai CD-ROM: £149.00

BASICUSSIONS
The Ultimate Percussion Tools! Percussion CDs & CD-ROMs available today are great but often of limited use. USB realized this and created a percussion CD with an angle : basic loops and tons of individual samples, all handily grouped by instrument : shaker, tamborine, triangle, congas, bongos, djembe. As the name implies, the loops are very basic but eminently usable, and are all available at several tempi. Likewise, the instrument selection concentrates on all the basic percussions that are present in virtually all music styles. Basicussions is a double CD, in mixed-mode AKAI/Audio. The first disc contains the percussion loops. The second contains the individual percussion hits. Each section is devoted to one instrument (e.g. in the shaker category, there are three different instruments over nine tempi). A generous bonus drum sounds section is located at the end of CD2. A must! Audio CD: £59.95, Akai CD-ROM: £149.00

SYNTH CITY VOLUMES 1,2 & 3
The Synth City collection will take you on a journey in the sonic world of the most famous synths on the planet. Each volume of Synth City is a double CD-ROM set chock-full of synth sounds, for the price of one. Superb sounds from the best synthesizers ever created, all ready to play from your sampler. Synth City Vol.1: JD-800 and M1 Impressions explores two best selling synths : the Roland JD800 (CD 1) & Korg T1 & M1 (CD 2). The subtle mix of warm analogue sounds & digital textures is typical of the JD800. The second CD features hundreds of ambient-oriented sound, with an emphasis on ethereal sounds & synth pads. 2 Akai CDROMs: £149.00. Synth City Vol.2: (Due out in June 98) Vector Impressions - discover the depths of the Korg WS & WS-AD (CD1) & of the Ensoniq VFX & SD1 (CD2). Synth City Vol.3: (Due out in September 98) The Virtual Analog CDROM (Nordlead, JP8000 & AN1X). Each volume: £149.00. Please call to check availability as not all volumes were released yet at time of going to print!

World Radio History

My Two Bits' Worth

BIT ONE/BIT 01/BIT 99 SYNTHS

GORDON REID is bitten by the memory of a brilliant, but deeply flawed trio of Italian synthesizer designs that could have been the last word in affordable analogue.

I remember the first time I played a Bit One. It was a hot summer's day in 1984, and my girlfriend and I had travelled to central London to buy a set of Chase CBP1 bass pedals. In many ways the poor man's Taurus pedals, these were made by an Italian company for Chase (a keyboard store more often known as the London Synthesiser Centre).

Soon after we arrived, Mr Pammi Singh, the store's manager and an industry 'character' without peer, calmly informed me that what I *really* needed was the CBP2 bass pedals because they sported a second oscillator, and for only a mere £100 more.

"Hold on", I said. "I can barely afford £199 for the standard model, let alone £299 for those." At which point, Mr Singh uttered the immortal line, "Ah, but with a girlfriend like that, how can you be poor?"

While I was battling with my better judgement and my cheque book, Mr Singh showed me a new keyboard that he had just received from Italy.

It was (in his words) a digital synth that, at £799, cost just half the price of a DX7, yet blew away the equivalent Rolands and Korgs of the day. It was black, sleek and, despite a five-octave keyboard, very compact. It was, he told me, warm, fat, flexible, expandable, and it had MIDI. It was a Bit One.

"Bit?" I asked. "Who are they?" In retrospect, it was hardly surprising that I had never heard the name, because it turned out to be the *nom de plume* of a far more famous manufacturer. Like the bass pedals that I had set out to buy, the Bit One was produced by Crumar, a company better known for its ghastly electric pianos, weedy string machines, and early 'multi-keyboards'.

Crumar had shot to prominence in the mid-'70s when a handful of bands such as Emerson Lake and Palmer, Greenslade, and The Enid adopted some of their quirky polyphonic ensembles. Unfortunately, these instruments seemed to have no place alongside the Moogs, ARPs and early Oberheims of the day. They even suffered in comparison with other Italian imports including Logan string ensembles and Elka Rhapsodies.

It was, nonetheless, an important company, if only because it made keyboards that were cheap enough for players who would otherwise have just been restricted to lusting after the unaffordable instruments of their heroes. Furthermore, the Crumar Multiman assured the brand's place in history, because it was the world's first 'multi-keyboard', a concept that was subsequently copied by ARP, Moog, Yamaha, Korg, and many others. But as the name Crumar always conjured up an image of

cheap shoddiness, serious players avoided their products, no matter how innovative or well designed they later proved to be (it's interesting to speculate how successful the Bob Moog-designed Spirit monosynth, or the Organiser T2 might have been had they been manufactured elsewhere).
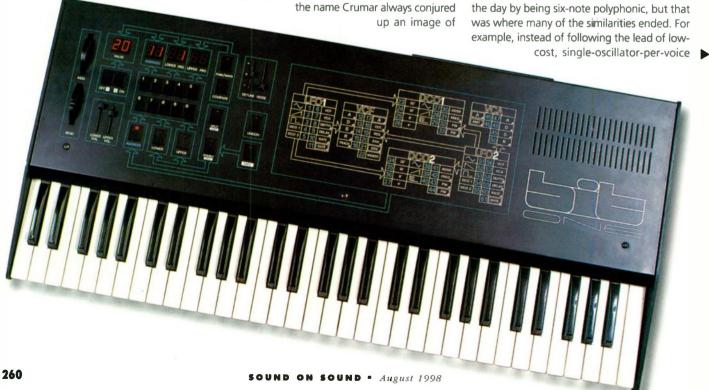
Clearly, Crumar understood this, so when

> **"A digital synth that, at £799, cost just half the price of a DX7, yet blew away the equivalent Rolands and Korgs of the day. It was black, sleek and, despite a five-octave keyboard, very compact."**

they designed their first MIDI keyboard, their name was changed to Bit.

THE TECHNOLOGY

The Bit One conformed to the standards of the day by being six-note polyphonic, but that was where many of the similarities ended. For example, instead of following the lead of low-cost, single-oscillator-per-voice ▶

synths such as the Korg PolySix and Roland Juno 106, its dual-oscillator voice structure was more akin to that of the Roland JX3P. And, whereas all these competitors offered just a single ADSR envelope per voice and a single LFO per patch, the Bit One featured two of each. What's more, you could assign these to many more destinations than the Roland or Korg models allowed.

Unlike any other low-cost synth of the day, the Bit One was also velocity sensitive. This facility

the whole keyboard. Instead of playing one patch in the Upper zone of the keyboard and one patch in the Lower, you heard the two superimposed upon each other. While this reduced the polyphony from six notes to three, the depth and complexity of the combined sounds more than compensated.

This may make the Bit One sound like some sort of low-budget super-synth, but in truth it wasn't. Yes, it had great strengths and, when you consider that its designer was Mario Maggi,

Another failing was the lack of option to modulate the pulse width via an LFO. Sure, you could vary the pulse width using an envelope or by velocity response, but you couldn't sweep it for those classic, lush PWM sounds. Even worse was the omission of fine-tuning and detune from the programming system. Instead of being programmable within each patch, you determined the detune between DCO1 and DCO2 by moving a slider on the top panel. So, if you wanted to jump between a thick ensemble patch and a thinner sound, you had to adjust this slider by hand as you played. Noise, while it was present on the synth, could only be accessed using a top panel slider. This was arcane. If you wanted a patch based purely on noise, you had to program it with both oscillators switched off, then introduce the noise from the front panel whenever you selected that patch. Weird, or what?

Finally, the MIDI specification was a joke, even in 1984. Compare it to its contemporary, the Juno 106 — a synth that understood all 16 MIDI channels, and transmitted and received note numbers, patch numbers, performance data, and any control panel changes. By contrast, the Bit One transmitted just note number and velocity on MIDI channel 1. When receiving (Omni mode only) it also recognised patch changes. And that was it! So, despite In, Out and Thru sockets, you couldn't use the Bit One to select patches on other keyboards or expanders, and you couldn't dump patches via ▶

> ## "I reckon that the Bits proved to be the right synthesizers at the right prices — but at the wrong time. The mid-'80s were the heyday of FM synthesis, and the world had temporarily tired of all things analogue."

was no afterthought, either: you could direct the velocity response independently to each of the LFOs, each of the DCOs' pulse widths, and to the attack times and envelope depths of both envelope generators, thus making both the VCF and the VCA touch sensitive.

The Bit One was also bi-timbral, and you could place any of its patches in Upper and Lower zones above and below a user-defined split point. There were even Upper and Lower outputs, so you could direct each zone to its own mixer channel for independent EQ and effects treatments. The Bit One had a duophonic Unison mode, and in the standard Play mode this stacked voices one to three under one key, and voices four to six under another, thus placing six oscillators under each finger. You could even select Unison in Split mode to get two different sounds — one created from the three voices in the Lower zone, and another from the three voices in the Upper zone.

If this wasn't enough, a Double mode allowed you to allocate two patches across

the man responsible for the Elka Synthex, this isn't too surprising. But its weaknesses were astounding. Minor complaints included the position of the Modulation and Pitch-bend wheels, which lurked one behind the other on the control panel itself. But far more serious was the inability to balance oscillator levels: DCO1 and DCO2 were always either 'on' or 'off'.

WHAT ARE THEY WORTH? — PART ONE

There were only three models of Bit, but I've separated the early Bit Ones from the later ones. This is because, for some people, the CEM filters make the later units less desirable. I don't think it makes any difference to their values, because the instruments' common failings are far more significant. If you get the chance (and the urge), buy a Bit 99.

MODEL	DATE OF RELEASE	PRICE ON RELEASE	SECOND-HAND BARGAIN	SECOND-HAND RIP-OFF
Bit One (CEM filters)*	Late 1984	£799	£100	£200
Bit 99	Late 1985	£699	£150	£300

* Apparently, the switch from the earlier to the later version occurred somewhere around serial number 000300. If the situation arises and you want to know which you have, you'll have to open the synth up and take a look.

▶ MIDI. If you wanted to create a patch library, you had to use cassette tape storage. Yurg!

THE NEXT GENERATION

In early 1985, Crumar addressed many of these problems by releasing an improved Bit synthesizer, though contrary to expectations, this was not the Bit Two keyboard, but the Bit 01 Expander module.

The deep, 3U-high Bit 01 (available in both black and ivory) was not just a repackaged Bit One, but very much a 'souped-up' version, with extra parameters and a dramatically improved MIDI specification. Additions included a parameter to control the maximum modulation depth applied by the Mod wheel, programmable noise on DCO1, programmable detune on DCO2, and programmable volume for the complete patch. Unfortunately, the Bit 01 lost the Bit One's ability to sweep the DCOs' frequencies using the VCA's envelope generator, and the Unison mode disappeared. On the other hand, the Bit 01 now allowed you to store 'splits' and 'doubles' as performances, with memory increased from the 63 single patches found on the Bit One to 99 patches/performances. It was a strange system though — 75 of the 01's memories were Bit One-style patches, and the other 24 were performances.

Equally important, the MIDI specification was much better. You could now leave Omni Mode, and the Bit 01 recognised all 16 MIDI channels. It also received controllers such as modulation and pitch-bend, making it much better for live work. Unfortunately, the 01 lacked a MIDI Out socket, so it was still impossible to save sounds over MIDI. Nevertheless, at £499, the 01 provided excellent value for money, and it should have been high on everybody's analogue shopping list.

THE FINAL CUT

Had Crumar released the Bit 01 as a keyboard, I might have gone elsewhere when, in the summer of 1985, I bought a Roland JX8P as my first MIDI keyboard. This boasted much the same specification as the Bit 01, but with

oscillator sync, an oscillator mixer, Roland's famous chorus effect, and pressure sensitivity. Three months later and (as far as I was concerned) with appalling timing, Crumar released the Bit 99.

In retrospect, the Bit 99 was simply a Bit 01 with a keyboard, performance controls, additional MIDI commands that the keyboard-less expander had not required, and the ability to chain three sets of 33 patches. This last attribute was an excellent feature, although overlooked by most players. The JX8P allowed me to chain eight patches but, in my view, the Bit 99's ability to step through 99 patches during a gig placed it in a league of its own. At its reduced recommended price of just £699, the Bit 99 should have been a winner.

RETROSPECTIVE

So what went wrong? The fact that the Bits were not resounding successes, and that Crumar disappeared just three years later, was nothing to do with the sound of the instruments themselves. Their lead synth patches and synth pads were remarkable, and they also excelled at punchy brass and polysynth patches. Indeed, many of the Bits' analogue sounds comfortably filled the shoes of the far more expensive Prophets and Oberheims. The Bits were also capable of some remarkably FM-esque sounds, and produced a wide range of DX-style electric pianos and harps.

Sure, they weren't perfect. For example, strange summing differences between DCO1 and DCO2 could give unexpected tonal responses whenever you played a key. Depending on your point of view, this was either a benefit, since it imparted a CS80-ish organic nature to many sounds, or it was a pain in the neck. Oddly, the differences did not seem to cycle as you played the six voices, so you could never be sure when it would occur, or by how much. There were other minor annoyances too, such as the (sometimes) audible stepping of the LFOs. But these deficiencies should not have

detracted from a warm and versatile family of synthesizers that could out-perform many more illustrious instruments.

Why then did the Bits disappear so quickly? Maybe it was the lack of tweakable knobs and switches. Maybe it was the lack of software support. Maybe, in the UK, it was the mistake of trying to sell them exclusively through one shop in central London.

I reckon that the Bits proved to be the right synthesizers at the right prices — but at the wrong time. The mid-'80s were the heyday of FM synthesis, and the world had temporarily tired of all things analogue. Digital effects units were just becoming affordable, and these were clearly superior to their analogue brethren. The new 'compact discs' were also digital, and the music industry was beginning to come to grips with ideas such as digital audio tape, hard disk recording, and techniques such as digital audio restoration. In this digital/digital/digital climate, the Bits were doomed, and their short reign came to an end.

But, even today, playing a Bit through a touch of chorus and a splash of reverb is a rewarding experience. And for those fortunate enough to be able to do so, layering two Bits through a stereo effects unit can produce stunning results. If you ever have the good fortune to get your hands on a pair of Bits (no giggling at the back please), you'll soon begin to wonder whether it's time to stop lusting after those ridiculously expensive and unreliable 'vintage' synths.

POSTSCRIPT

Incidentally, I had no choice in the end but to buy the CBP2s from Mr Singh and, ultimately, it proved to be the right decision. Unfortunately, I sold them in 1989 for just £99 — that was a big mistake. If they (or another set tucked away in somebody's loft) are still in the near-mint condition that mine were when I last saw them, I would gladly buy them back. Contact me, and you'll be guaranteed a warm reception. [SOS]

WHAT ARE THEY WORTH? — PART TWO

Just for comparison, here is a list of prices (all obtained from *SOS* June '98) that owners are charging for some of the other synths mentioned in this article. It makes you worry about the buyers' sanity.

MODEL	PRICES BEING ASKED...AND PAID!
Elka Synthex	£1,250
Korg Polysix	£300
Oberheim OB8	£699
Sequential Prophet 5	£1,099
Roland Juno 106	£550
Roland Jupiter 8	£999
Roland JX3P	£350

Korg TR-Rack

Korg's hugely successful TR-Rack is a sample-based subtractive synthesis module derived from their immensely powerful Trinity Workstation (see the *SOS* two-part review in December '95 and January '96 issues). The TR-Rack (see SOS February 1998 for the full review), costs just £999, and contains the Trinity's ACCESS (Advanced Control Combined Synthesis System) synthesis technology coupled with its impressive multi-effects and multi-samples. However, although the TR-Rack is obviously a close relation of the Workstation, it offers a few features that even the Trinity itself didn't have. The Rack comes with 32Mb of sound ROM (the Trinity weighed in with 24Mb), and has twice as many built-in programs. Another thing the TR-Rack is not short on is effect options with a master bank of 14 effects and a further 100 available as inserts. Up to 8 insert effects and 2 master effects can be used at once. In terms of sounds, the Rack offers a healthy

512 programs and 512 combinations, and provides no fewer than 24 drum kits. What's more, all the multi-samples have been sampled at 48kHz ensuring excellent quality.

The TR-Rack is a good buy for its many quality effects and multi-samples alone, but it also offers very extensive programming options for those who take their sound design a little more seriously.

Two multi-samples can be assigned to an oscillator, and two oscillators, each with its own filter and amplifier, can be used together to create a program. Furthermore, it's possible to combine eight programs in velocity and key split/layer textures. Another mode called 'Multi' provides 16-part multitimbrality for sequencer work.

Full editing and programming is made possible through the software packages supplied free with the TR-Rack, including Emagic *SoundDiver* Trinity editor/librarian and Korg MIDI driver software. By using the serial port on the rear panel, the TR-Rack can be connected to Mac or PC without the need for a MIDI interface and both PC and Mac versions of the software are available to support whichever system you use.

Fortunately for one lucky winner this month, *SOS*, in association with Korg UK, are giving away a TR-Rack as a competition prize, so let's cut to the chase and discuss how you can go about becoming that person. All you have to do is select the appropriate answers to our mind-stretching questions below, answer the tie-breaker and then post the completed entry to the address below. Simple.

the small print

QUESTIONS

1. Which one of the following describes the synthesis method used by the TR-Rack?
a. VISA ☐
b. ACCESS ☐
c. DELTA ☐
d American Excess ☐

2. The TR-Rack has a huge cache of sound ROM. How much?
a. 32Terabytes ☐
b. 32Bytes ☐
c. 32Dogbytes ☐
d. 32Mbytes ☐

3. If you want to edit/program your TR-Rack from an external computer you will need to connect the two together, but by which of the following methods?
a. By using the serial port on the rear panel ☐
b. Using jump leads ☐
c. By holding it in front of the monitor and pressing the Return key ☐
d. By satellite uplink ☐

Would you like to receive more information on Korg products? If not, please tick this box ☐

Post your entry to: SOS Korg TR-Rack Competition, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ.

Korg TR-Rack Competition

TIE BREAKER

Imagine that the TR-Rack's 100 effects inserts and 14 master effects still don't fulfil your recording needs. What unusual additional effect would you like to see and why? Answers should be a tidy 30 words or less.

Name ....
Address ....
Daytime tel. no. ....

WELCOME TO THE CLUB 2000

102 £276.00

142 £323.00

182 £410.00

Imagine you're a real hot signal from out of town.

When you check into the grand hotel of sound you'll soon discover the CLUB2000 right next to the lobby. Come on in and grab a seat in the high head room, where even the hottest signals can chill out and relax before attending the evening festivities. Once you're inside you'll have to be discrete, just like the ultra quiet microphone pre-amps and balanced line inputs at the CLUB2000.

The evening festivities will include the CLUB2000's powerful 3 band channel EQ with mid sweep, which gives you more elbow room when all those signals start to dance - more than any other mixer in it's class. Other late night features include: 2 stereo line channels and 2 band EQ with a twist. 2 (pre post) auxiliary sends, PFL listen per channel, 48v phantom power (for your condenser microphone buddies) and smooth 60mm faders for a precise "attitude" adjustment.

As you begin to work the room and things start to heat up, you'll meet up with your old friends: L/R inserts and the 2 master auxiliary sends with a stereo return ticket. When you finally leave the CLUB2000 you can rest easy, knowing that you're completely balanced and ready for the trip home.

AND... if that's not enough, we'll introduce you to three great looking models (102,142 and 182) to help convince you to stay!

APPLICATIONS

- Public Address.
- Bands, Theatres.
- Installations, Conferences.
- Clubs, Schools, Colleges.
- Places of Worship etc.
- Recording.
- Computer Soundcards.
- Video Audio.
- DJ mixing.
- Keyboards etc.

KEY FEATURES

- All inputs balanced.
- Renowned "British" Eq.
- CD Audio Quality.
- Two stereo channels with Eq.
- Main outputs balanced (+28dBu Max).
- Compact, low noise design.
- Internal power supply.
- Rack kit supplied (102 & 142).
- Internally modular.
- Robust, all steel construction.

WHEN YOU NEED TO GET SERIOUS....

Studiomaster

STUDIOMASTER HOUSE, CHAUL END LANE, LUTON, BEDS, LU4 8EZ ENGLAND.
TEL +44 (0) 1582 570370 FAX +44 (0) 1582 494343
STUDIOMASTER INC, 3941 MIRALOMA AVENUE, ANAHEIM, CALIFORNIA 92807 U.S.A.
TEL (714) 524 2227 FAX (714) 524 5096
WEB SITE http://www.studiomaster.com

Please send a full colour Club2000 brochure to:

Name

Address

Postcode

Tick here if you would like to be kept up to date with the Studiomaster product range.

World Radio History

World Radio History

TB303 with modification, £750; TR606, £100; DX7 with cards, £250; DX100, £100, SK30, £275. ☎ 01670 523363 (Northumberland).
ROLAND JUNO JSQ60 synth, boxed, sequencer, excellent condition, £400 ono ☎ 01722 744196 (Wilts).
ROLAND JUPITER 4 good condition, £250; Hohner HS2E FM multitimbral synth, good sub bass, 16-note poly, very good condition, £150. ☎ Justin 0171 686 4234.
ROLAND JUPITER 6 analogue synth, manual, all original and in superb condition, hard case, stand, £450 ono. ☎ Pete 0118 941 9372 (Reading).
ROLAND JUPITER 8, mint, Robin Whittle MIDI retro-fit, expansion memory module, FM filter, pro flightcase, studio use only, £1250 ☎ Marc 07050 278508.
ROLAND JV90, 2 expansion-cards, home use only, immaculate condition, boxed, video, manuals, £800 ☎ Paul (evenings) 01626 863541 (Devon).
ROLAND JV880, £300; Korg 05R/W, £200; Akai SG01V, £100; no offers, have all been racked from new. ☎ Justin 01704 506151 (Merseyside).
ROLAND JV1080 with dance board, Akai S2000, Novation Bass Station rack, Alesis Quadraverb, Alesis MMT8, Spirit Folio 12:2 desk, Roland PC200 mother keyboard, all as new, £2500 ono. ☎ 01789 293824 (Warwickshire).
ROLAND JX3P keyboard, very good condition, manual £195. ☎ 01206 397217 (Colchester).
ROLAND JX8P, classic analogue synth, excellent condition, manual, heavy-duty flightcase, 2 memory cartridges, still boxed; Atari Editor, velocity sensitive, aftertouch over MIDI. ☎ 0191 372 2297 (Durham).
ROLAND JX8P with PG800 programmer, classic analogue with flightcase, £290, no offers ☎ 0181 592 2410.
ROLAND MKS 7, modular equivalent of a three-part multitimbral Juno 106 and TR707, 6 outputs, responds to slider movements, £350; Roland PG300 programmer for Roland Alpha Juno 1, 2 or MKS50, £200. ☎ 0141 638 6687 (Glasgow).
ROLAND MKS70, with PG800 controller, £600; Akai S900, £250; Quadraverb, £200; Poly 800, £120; Casio CZ101, £110; TQ5 tone module, £150; 626 drum machine, £100 ☎ Toby 0171 729 4183.
ROLAND MKS80, £750; Alesis QS6 with dance expansion card, £480; Juno 1, £230, PG800, £170, Boss DR5, £180; Wavestation with 2 cards, £450; LXP1, £180; DMT8, £580, all as good as new; MPX1 and JP8000 wanted. ☎ Steve (after 7pm) 0151 727 0110 (Lancs).
ROLAND MV30 workstation, £275; Studiomaster 24-channel mixer, £450; M EQ 230, mint condition, £150; Frontline 8-channel rackmount mixer, £100; Atari 1040FM with monitor, £125. ☎ 01375 387823 (Essex).
ROLAND P55 piano module, 3-part multitimbral, reverb/chorus effects, immaculate, £250 ono ☎ Paul 01287 638018 (Cleveland).
ROLAND RS09 organ/strings, full working order, suit collector, learner, £70 ono. ☎ 0181 740 8916.
ROLAND SC55 sound canvas, manual, remote control, power suppy, good condition, £260 ono. ☎ Dave (after 6pm) 01635 528905 (Berkshire).
ROLAND SC55 MKII Sound Canvas, 28-note polyphonic, 354 sounds, 9 dynamic sounding drum sets, £320 ono, or swap for Akai S900/950 sampler. ☎ 01523 160361 (West Midlands).
ROLAND SCP55 sound canvas with MCB3 interface, brand new, never used, cost £345, will accept £245 ono ☎ 0121 705 1286.
ROLAND SC88, boxed, manual, £380 ono; Yamaha DX7, £275 ono; Emu Proteus power supply PCB, £50; dead synths: Ensoniq Mirage, Sequential Circuits Pro 1, offers considered. ☎ Steve 01626 331390 (Devon).
ROLAND SH09, £249; SCI Pro 1, £449; SH3a, £199, Roland MC202, £265; TB303, £699; MKS70, £449; HP310 printer for Mac, £49 drumtraks, £199; Oberheim DX, £199. ☎ Tom 01384 353694.
ROLAND SH101, boxed, £320; Silver TB303 case, £25; original manuals for TR909, TR808 and TB303 for £20 each; manuals for MC202, Juno 106, £15 each, Korg MS20 set, £25, MS10, £15. Mark. ☎ 01226 206767 (Barnsley).
ROLAND U20 keyboard, £300, or swap for Digitech Vocalist with MIDI; Atari Falcon Mk1 hard disk recorder, 350Mb hard drive, 4Mb RAM, £300 including CD-ROM software. ☎ Martin 0116 2554285 (Leicester).
ROLAND U110 with card, £150; Roland D110, £120; Yamaha TX81Z, £100; Yamaha EMT10 piano, £90; LA Audio 16-channel gate with MIDI, £120. ☎ Danny 01294 215529 (Ayrshire).
ROLAND U220 module, £220, U110 PCM

data ROMs, £25 each; Aphex 109 parametric EQ, £275; Midiverb 2 effects, £140; Cubase Score v3 with dongle, for the Atari, £200; Award MB10 DI box, £20. ☎ 0181 902 9784.
ROLAND W30, 16-track sequencer, sampler and keyboard, comprehensive disk library, hardly used, boxed with instructions and stand, £550. ☎ 01949 861891 (Nottingham).
ROLAND XP10, as new, 5 months old, hardly used, boxed with manuals, genuine reason for sale, £300; Roland SH101 with mod grip, grey, very good condition, £250. ☎ Steve 0181 406 7764/343 3917.
ROLAND XP50 workstation, flightcase, in very good condition, £695. ☎ 01473 219603 (Ipswich).
ROLAND XP80, new, £985, Alesis QS6 with dance expansion card, £420; Roland Jupiter 8, MIDI, £780; PG800 with lead, £170; Sony HRMP5 pro effects, £280; all ono and as new. ☎ 01523 172676.
ROLAND XP80 plus expansion, 6 months old, manuals training video, home use only, £795. ☎ 01485 535079 (Hunstanton).
SEQUENTIAL PRO ONE with Kenton CV converter, £425; Yamaha TX81Z, £120; Cubase VST 3.552 for PC, £200; all mint condition. ☎ 0115 993 0473 (Nottingham).
SEQUENTIAL PRO ONE immaculate condition, £300; Fostex recorder, 250, £250; Galanti combo piano, £195; Atari 1040 ST, SH204, two SF354s, Cubase 3.0, SM124, £300. ☎ 0141 942 4120 (Glasgow).
SEQUENTIAL SIXTRAK analogue synth, arpeggiator, sequencer, case, £295 ono; Studio Vision Pro 3.02 software, sequencer, Galaxy librarian, studio 4 8x8 patchbay, £395 ono. ☎ 01524 736158.
SOLTON MS50 intelligent keyboard, stunning sounds, programmable styles, £850; Ensoniq VFX/SD synth, great atmospheric sounds, £295; Roland RA50 intelligent module, £195; all ono. ☎ 0181 202 5353.
TECHNICS KN800 synth, speakers, rhythms, effects, PCM sound, £350; Roland U110 module, £150; Roland MS1 sampler, £180; rack trolley, ideal for E16, £90. ☎ 0171 627 5474.
TECHNICS KN2000 semi-pro keyboard, 16-track sequencer, realistic piano and wind sounds, manuals, disks, softcase. ☎ Martin or James (after 5pm) 01452 305187 (Gloucester).
TECHNICS KN3000, home use only, in very good condition, boxed, £899; Fostex XR5 4-track cassette, £150. ☎ 01394 626900/01206 263250 (Colchester).
WALDORF MICROWAVE XT, used once, start the bidding at £1100, best offer secures sale. ☎ 01792 532915 (Kent).
YAMAHA AN1X, boxed, manuals, £650; MC303, £350; Zoom 1204 effects, £110; buyer collects. ☎ David 01943 873711 (Leeds).
YAMAHA AN1X, 8 months old, excellent condition, PSU, £495. ☎ 0171 843 4410/0181 656 3634 (Croydon).
YAMAHA CS1X, £330; Roland MC303, £330, both boxed, manuals, excellent condition, light studio use. ☎ 01628 823842/0498 695215 (Berkshire).
YAMAHA CS1X, boxed as new, includes computer interface, lead, editors, hundreds of new patches, £400 ono. ☎ Paul 0181 554 9735.
YAMAHA CS1X, immaculate condition, free pair of JVC speakers, £365 ovno. ☎ Wayne 01782 205842 (Staffs).
YAMAHA CS15, analogue, monophonic, 2xVCOs, VCFs and ADSRs, high, band and low-pass filters, 12/24dB filters, portmento, pulse-width modulation, glide, detune, mint condition with manuals, £330. ☎ 01296 334866/0374 963829 (Bucks).
YAMAHA CS30 analogue beast, rare, £350; Korg KPR77 analogue DMC, £75; Alesis Micro Limiter, £60; Boss compressor/limiter/noise gate, £50; Yamaha QX5 **sequencer**, £50; Cutec 10-band graphic, £20. ☎ 01428 642070 (Surrey).
YAMAHA DX7II FD, disk drive, storage, hard case, excellent condition, top-of-the-range classic synth, manual included, £300 ono. ☎ Kenn 0131 476 3699 (Edinburgh).
YAMAHA DX21 FM synth, manual, excellent condition, £140 ono. ☎ Gary 0121 445 5274 (Worcs).
YAMAHA EX5, under 1 month old, £1600; Alesis QSR, mint, £400; Korg NS5R, mint, £300, all home use only, will haggle. ☎ Bryan (after 5pm) 01387 750427 (Scotland).
YAMAHA GS2 digital keyboard, early FM synthesis, 73-note weighted keys, sounds are loaded with magnetic data-strips, good library of sounds, very collectable, can be MIDI retro-fitted, offers rare 1950 Selmer Clavioline, in good condition, £225; EMS Vocoder 1000, circa 1976, used by Herbie Hancock, excellent condition, offers. ☎ 01942 814818.
YAMAHA MU100 sound module, over 1700 voices, 46 drum-kits, 64-note polyphonic, 32-part multitimbral, vocal harmoniser, mint

condition, £425. ☎ Roy 01384 294026 (West Midlands).
YAMAHA PFP100 piano, complete with stand, excellent condition, 88 notes, weighted action, ideal as mother keyboard or stage piano, home use only, £750. ☎ 01827 715480 (Warwickshire).
YAMAHA PSR8000, new model, microphone input, choir harmoniser, sampler, disk drives, sequencer, screen lyrics, 6 weeks old, warranty, £1350. ☎ 0976 902529 (Weymouth).
YAMAHA SY77, 2 soundcards, mint, £460; Five-star flightcase for M1, new, £75; M1 Valhalla gold gards B101 and B103, £25 each. ☎ Peter 0181 675 0651.
YAMAHA SY85 workstation synth, 30-voice poly, 61-note keyboard, disk drive, 8-track sequencer plus drums, 16-part multitimbral, sample RAM expanded to 2.5Mb, 256 voices/128 performances, boxed, manuals, extra sounds, latest OS, good condition, one owner from new, £725, ono. ☎ 01354 695239.
YAMAHA SY85 workstation with Calsbro Colt 65W amp, also includes extra sound disks, labels, stand and manuals; all in excellent condition, £720 ono. ☎ Ollie 01235 763733 (Oxon).
YAMAHA SY85 workstation, stand, cards, memory expansion, £550 ono. ☎ Martin 01628 473076 (Bucks).
YAMAHA SY85 keyboard/workstation, 16-track sequencer, effects processor, disk drive, all manuals, still boxed, £600. ☎ 01858 466948 (Leicestershire).
YAMAHA SY85 expanded, 2.5Mb, disks, Atari ST sample software, Cubase utilities, complete dance studio, mint condition, £650, no offers. ☎ Dave 0161 308 3202 (Lancs).
YAMAHA SY99 76-note classic, in very good condition, case, manuals, extra sounds, £750 ono; Atari 1040STF, 4Mb RAM, high-res monitor, Cubase v2, leads, £150. ☎ Jon 01566 784665 (Cornwall).
YAMAHA TG100 tone generator, 16-bit, multitimbral, MIDI, £150; Philips digital compact cassette recorder and tapes, £140; Boss DR550 drum machine, £70; swaps considered. ☎ Dave 01189 782017 (Wokingham).
YAMAHA TX802, multi-DX7 expander, £195. ☎ Alan 01494 677106 (Bucks).
YAMAHA TX 81Z FM 8-part multitimbral sound module, immaculate, £90; very basic drum machine, analogue delay unit, flanger, all in good condition, £40 for the lot. ☎ Mark 01872 863966 (Cornwall).
YAMAHA YS200 workstation, 100 sounds, 100 user memories, 8000 events, sequencer, soft bag, Yamaha RX21 drum machine, mint condition, £185 ono. ☎ 0181 449 0217.
YAMAHA VL7, mint condition, £600 plus carriage; Roland EP3 digital piano, £100; Atari STE, 2.5Mb; Atari monachrome monitor, Steinberg and Office software, extras, £300. ☎ Lee 01291 627653.

RECORDING

ACOUSTIC 3311 studio monitors, brand new, professional or home use, cost £1785, will sell for £1500 ono. ☎ Rob 01483 300178 (Surrey).
ACOUSTIC 3311 studio monitor, just the one, in pristine condition, used once, boxed as new, includes a 12-inch polywoofer, 5-inch midrange and 3-inch tweeter, frequency controls, £750 ono. ☎ 01455 233371 (Leicestershire).
AKAI MG1214, 12-track studio with tapes, £899; Korg digital mixer, 168RC, £575 new, ADAT LX20, boxed, £1299; Alesis Microverb 4, £80; Nanoverb, £60. ☎ 01273 686637 (Brighton).
ALESIS ADATS, two of them with BRC and v4.03; Soundtracs Megas desk, 26 mono, 6 stereo, 24 subgroups, 6 aux, looms, excellent condition, great package, £5000. ☎ 0836 344975/01225 471115 (Bath).
ALESIS ADAT v4, 2700 hours use, well maintained, includes EDAC, loom and 25 used tapes, £700; JL Cooper Datasync, £175; XR300 sync, £100; 18-inch scoop bins, £250. ☎ 01873 860316 (Herefordshire).
ALESIS ADAT XT, as new, boxed with blank tapes, very small amount of home use, £1650. ☎ 01925 740335 (Cheshire).
ALESIS ADAT XT, immaculate, 170 hours, home use only, £950 ono; Yamaha Pro Mix 01 wanted, cash waiting. ☎ 0151 643 1964 (Wirral)
ALESIS COMPRESSOR, £175; Alesis 1622 mixer, £200; Behringer Exciter, £100; Yamaha full-size electric piano, £700; all ono. ☎ Andy 0121 449 8024 (Birmingham).
ALESIS MICRO GATES 3-in-one rack, £75 each or £195 together; MXR dual 16-band graphic EQ, £135; all in excellent condition. ☎ 01865 776587 (Oxford).
ALESIS MIDIVERB 4, £145; Samson MPL2242, £285; Alesis Monitor Ones, £155;

World Radio History

Samson Servo 150, £105, Aiwa ADF450 cassette, £45; all mint, hardly used, £675 for the lot. ☎ Andy 01253 890048 (Lancs).

ALESIS 1622, £350; Alesis RA100, £175; Microverb 3, £60; Fostex M80, looms, £325; Fostex M20 2-track, £250; Fostex 4030, £150; Yamaha NS10M, £175. ☎ Rory 0181 870 1529.

ALESIS MULTIVERB II, £190; Art Multiverb II, £170; Emu Proteus, £200; two Alesis ADATs v4.03, £600 each, Alesis BRC v4.03, £750; Soundtracs Megas, 20 mono, 6 stereo, 24 sub, 6 aux. ☎ 01225 471115/0836 344975 (Bath).

ALESIS QUADRAVERB 2, £195, Terratec Sound Base 1 soundcard, £15, Midiman Portman PC/S MIDI adaptor, £20. ☎ Ian 01484 646242 (Huddersfield).

ALLEN & HEATH GS1 mixer, 16:8, MIDI automation, in very good condition, £450; Tascam MM1 keyboard, mixer, 20-channel, £250 ono. ☎ Tim 01483 452400 (Guildford).

ALLEN & HEATH SABRE mixing desk, 24:16.16, 40 channels with EQ and MIDI mutes on mixdown, meter bridge, stand, good condition, home use only, £2850 ono; Rane SM26, mixer, £200 ono; Tascam MSR24, good condition with remote, £3850 ono. ☎ 01604 588618 (Northampton).

ALLEN & HEATH SABRE recording console, 24:8, SPA with VU meters, small studio, manual, cash only sale, £2000 ono. ☎ 0181 346 6337

ALLEN & HEATH S2 desk, 16:8:16:2, 40 inputs on mixdown, 4 aux, 3-band sweepable EQ, automated MIDI muting, serviced, manuals, home use only, £750 ono. ☎ 0121 442 4326 (Birmingham).

AMPEX 16-TRACK 2-inch, very good condition, full remote, tape counter, loom, £1800 ono. ☎ 0118 961 7660 (Berks).

ART MULTIVERB LTX 16-bit digital effects unit, £100; Valley Dynamite gate/compressor/ limiter, dual channel, £140; OHM MR228 PA speakers, 300W RMS, 80HZ-17kHz, £330. ☎ 01460 68157 (Somerset).

ART TUBE mic preamp, £100; Fostex compressor, £35; Alesis SR16 drum machine, £150; JL Cooper PPS100, SMPTE synchroniser, £110 ☎ Dee 0171 701 3883/0956 621669.

BEHRINGER EURODESK, immaculate, £800, TL Audio Indigo valve compressor, £230; BBE 362 Sonic Maximiser, £120; ART tube MP, £75; Audio Technica 4033a condenser microphone, £210; all in very good condition ☎ Gary 01278 782349 (Somerset).

BEHRINGER EURODESK, 24:8, over 60 inputs on mixdown, boxed, manuals, home use only, £1100, Emu Procussion, 6 separate outs, huge number of drum sounds, £240. ☎ Mark 0181 847 2135 (Brentford).

ENSONIQ DP2, lovely effects, loads of detail, £400, swaps considered. ☎ 0171 736 8823.

EVENT 20/20 MONITORS, 200W RMS, 8-inch woofers, 1-inch tweeter, mint condition, hardly used, £765 including full flightcase. ☎ 01895 237292/0370 413420

FERROGRAPH LOGIC 7 high-speed recording machine, 15ips noisy, new re-lapped heads, tech manual, £175 plus carriage; Tanberg TCD 310 Mk2 cassette deck, as new, £80 plus carriage ☎ 0161 6537914.

FOSTEX B16 analogue half-inch, multitrack, good condition, looms, manual, Ampex tapes, £800; Neumann U87 mic with suspension, £850. ☎ Bristol 0117 942 9490 (Bristol).

FOSTEX B16 half-inch multitrack, good condition, well looked-after, £1000 each; Seck 18:8:2 mixer, £200. ☎ Richard 0181 960 4466.

FOSTEX B16, 16-track half-inch recorder, excellent condition, £750. ☎ 01538 702259 (North Staffs).

FOSTEX D80S, two of them with 1.3Gb hard drive, immaculate £820; Spirit studio 16:8:2, £730; Yamaha SY85, £530; Sony HRMPS effects, £200; 2 Behringer Autocoms, £110 each; Sony DTC550 DAT, £265, Atari 1040 STE with SM124 monitor, £130. ☎ Carl 0114 250 0991 (Sheffield).

FOSTEX D80, extra recording time, caddy, £1100 ono; 8-way headphone splitter, £30. ☎ 01722 744196 (Wilts).

FOSTEX D80 hard disk 8-track, Folio F1 16-channel desk, looms, Art FX1, Zoom 1201 effects, complete set-up worth £1700, will sell for £1200 ono. ☎ Johnathan 0114 255 7723/0171 681 9222

FOSTEX DMT8 hard disk 8-track recorder, immaculate, boxed with manual, £850. ☎ Graham 01423 771440.

FOSTEX M80, 8-track, £349; Fostex 812 multitrack mixer, MIDI mute, £249; Yamaha MS5 programmable timecode, MIDI synchroniser, £39 or £550 for all of the above. ☎ 01494 677106 (Bucks).

FOSTEX E16, remote metering, autolocate, 4050, £400; Studiomaster 16:8:16, £400; Vortexion valve mixer, 4:4:1 rare 1950s, £400; pair of Neve EQs, weird old compressors. ☎ Paul 01462 626711 (Herts).

FOSTEX M80 8-track analogue recorder and looms, Roland 626 drum machine with tape sync, £400. ☎ Robert 01772 784788 (Lancashire).

FOSTEX R8, MTC, footswitch, Fostex 812, mixer, MIDI mute, looms, six reels, all in excellent condition, boxed with manuals, will split, £950. ☎ 0121 743 9054 (Birmingham).

FOSTEX R8 8-track recorder, remote panel, only £400 ono. ☎ 0121 449 9176 (Birmingham).

FOSTEX R8, immaculate condition, with 2 8-way looms, footswitch, manual, new tape, £480; or with JL Cooper PPS2 SMPTE/smart FSK synchroniser, £550. ☎ Ian 0181 530 7988.

FOSTEX R8, remote extension, 7 tapes, demag, boxed, manuals, home use only, £600 ono; Alesis Quadraverb, boxed, manuals, home use, £200 ono. ☎ 01483 221243.

FOSTEX R8, TSI sync, looms, tapes, £575; Fostex 812 mixer, £300; Yamaha EME reverb, £50; Yamaha EMT sound module, £50; home use only. ☎ Clive 01243 572633 (West Sussex).

FOSTEX R8, perfect condition, with new tapes, looms, JL Cooper PPS1 sync box, £600 ono; Alesis MEQ 230 stereo graphic EQ, £120 ono; buyer collects. ☎ 0141 420 1673 (Glasgow).

FOSTEX X26, 4-track cassette recorder, hardly used, boxed with manuals, £120. ☎ Roy (after 6pm) 01535 665299 (West Yorkshire).

GENERAL MUSIC 8:2 powered mixer, 30W per side, on-board effects, as new, cost £700, will sell for £350. ☎ 01933 275690 (Northants).

HOME STUDIO CLEARANCE, mixers, modules, recorders, effects, looms, tapes and more. ☎ Terry 01420 84030 (Hampshire).

JBL EON 15 powered speakers, suitable for front-of-house or monitors, £750. ☎ Rachael 01903 733061 (Littlehampton).

JBL SPEAKERS, 450 Watt PA, 15-inch cones and treble horn, as new, £100 for the pair; one 225 Watt trapezoid floor monitor, as new, £250. ☎ Mark 01326 376674 (Cornwall).

JOEMEEK VC3 Pro Channel, superb mic preamp/compressor/enhancer combo with phantom power, little used, boxed, manual, mint, with PSU, superb sound, add magic to your vocals, £165 ☎ 01354 695239.

KORG SE 500 range echo, full spec, very similar to Roland RE series, includes cover and circuit diagram, £275. ☎ Paul 01204 593301 (Lancs).

LEXICON LXP1 reverb/effects, superb condition, 2 units, £225 each; Lexicon MRC MIDI remote control with digital readout, controls most Lexicon units and most MIDI gear, £225; Roland GM70 guitar MIDI converter and pickup, top-of-the-range, rackmount, 128 memories, plays any MIDI synth/sound module direct from guitar, immaculate condition, £295. ☎ 01482 448767.

MACKIE 32-CHANNEL 8 buss mixer with stand and manual, immaculate condition, £2500 ono. ☎ 0171 381 0747.

MACKIE 1604 mixer, £400; Alesis Midiverb, Midifex, Microverb, Micro-enhancer, £200 or £500 for the lot. ☎ 01427 811434 (Lincs).

MACKIE CR1604 mixer, 16 inputs, two stereo outs, great EQ, 6 aux sends, insert points, multitrack possible, rack ears and optional Rotopod for rotating jackfield to most convenient position, boxed, manual, good condition, sounds great, one owner from new, £620, ono. Not, repeat not, VLZ version. ☎ 01354 695239.

MACKIE 1604 mixer, 16 channels, 3-band EQ, 6 aux, 8 insers, perfect condition, home use only, £550; Atari STE with 14-inch high-res monitor, Cubase v3.1, all leads, mouse included, £250; PCI Pentium, 200MMX, 64Mb ROM, 2.3Gb hard drive, MIDI card, Event Darla soundcard, £1150. ☎ 01788 860357 (Leicestershire).

MACKIE 1604 VLZ mixer, mint condition, hardly used, boxed as new with instructions, £590; two dbx noise gates, £30 each or £50 for the pair; can deliver. ☎ 0121 359 1232/0958 287231 (Birmingham).

MACKIE 48-CHANNEL console, meter bridge, stand immaculate, £4000 ono. ☎ 01924 262544/0410 986368.

MARTIN AUDIO MONITORS, £275; Alesis Datadisk midifile player, £195, JV world board, £150; professional light stands, winch type with castors, £150. ☎ 01604 770660 (Northants).

MTR 6-CHANNEL stereo headphone amp for studio monitoring, brand new, £200; flightcase for Spirit Studio 16:8:2 or similar sized desk, £200; Pro racks/flightcase, 20 units with 8 castors and full suspension, brand new suitable for professional touring band or mobile studio set-up, £350. ☎ 01434 685358 (Northumberland).

NEUMANN U67 valve microphone, power pack, £1000 ☎ 0181 449 6110 (Herts).

OTARI MASS quarter-inch mastering machine, excellent condition, £500 ono, Calrec condenser mic, £120; part-exchange possible. ☎ Joe 0181 881 2262.

PEAVEY HISYS MXT, monitors, 15-inch speakers with horn, excellent, very loud, £230 each, also, older style, £150 each; Peavey powered 15-inch plus horn, £300. ☎ Dale 0117 985 9494 (Bristol).

PEAVEY HI-SYS 3S speakers for sale, 700W per channel, good condition, £650 ono; Atari STE with Cubase program, £150. ☎ Mat 07970 385122/0181 300 5334 (Kent).

PHILIPS CDR870 CD-R and CD-RW recorder, brand new, unwanted gift, includes 4 CD-R disks and one CD-RW disk, was £499 new, will accept, £399. ☎ Andrew 0161 257 3798 (Manchester).

PRO STUDIO CLEAROUT, Korg Z1 expanded, Mutator, Yamaha O1V mixer, expanded, Wavestation EX with 6 cards; ESI 4000, Beat Extractor, QY700, Eventide 3000 DSE with chips, loads more, offers or swaps. ☎ 0181 810 5370/248 3391.

REVOX A77 reel-to-reel, manual, 3.75/7.5ips, £200. ☎ Peter 0118 978 2709 (Wokingham).

REVOX A700 recorder, 2 tracks, 2 AKG D224C microphones, 2 Beyerdynamic M260N, Revox M3500 studio-quality recording, 60 reels of professional recording tape, 10.5-inch Nab spools. ☎ 01525 406092.

ROLAND DM800 8-track hard disk recorder, 1Gb external storage, never crashes, video output, PC keyboard included, £1100 ono. ☎ 0171 935 5234.

ROLAND SRE555 chorus echo, rackmount, £500; SCI six-track analogue synth, £350; Hammond analogue drum, £85; Korg Minipops analogue drum, £90. ☎ 0171 274 7044.

ROLAND VS840, 8-track digital workstation, 4 weeks old, £750 or swap for Fostex D80/D90. ☎ Brad 01633 865758 (Gwent).

ROLAND VS880 V-Expanded digital multitrack recorder, fully expanded, built-in effects board, tutorial video, manual, absolutely immaculate condition, boxed, £1100. ☎ 0958 607578 (West Midlands).

ROLAND VS880, enhanced with effects board, 1Gb internal drive and Roland case, perfect condition, £900. ☎ 01749 673464 (Somerset).

ROLAND VS880 with effects board, £1050; Yamaha QY700, hardly used, £650; Sony A6 DAT, £400. ☎ Andy 01992 560341 (Essex).

ROLAND VS880 with SCSI, jaz drive, 2x1Gb, disks, effects-board, mint condition, £1250, Drawmer LX20 compressor, as new, £100. ☎ Paul 01926 316607 (Warwick).

SANSUI MR6 classic 6-track cassette recorder, 3U rack mount, 50% more tracks than your average 4-track, bounce in stereo, recent full service fom Panic Music so sonically and operationally as good as new, double speed, varispeed, Dolby C, sync track, great condition, box, manual, excellent sound, £365, ono. ☎ 01354 695239.

SECK 18:8:2 mixer, manual, boxed, very good condition, home use, £525. ☎ 0171 833 9751.

SECK 18:8:2, mixing desk, good sound, complete with power supply and manual, good condition, £450 ☎ Stuart 01353 669854 (Cambs).

SONY DTC DAT 690, home use only, manual, £350; Atari Mega 2, 2Mb high-res monitor, software, Cubase, 1 MIDI in, 2 out, box, keyboard, mouse, all for £250; Roland VS1 vintage sound module, £220 ono. ☎ 0171 967 9693.

SONY PTC1000 ES, full-size professional DAT machine with XLRs, copyright, error status, EBU lock, Audio Design professional expansion pack, mint condition, £395 or £475 with 30 tapes. ☎ 0171 681 8990.

SONY TCD D7 portable DAT, machine, excellent condition, no power supply, uses batteries, £250; Roland DJ70 sampler workstation, mint condition, no manual, £250 ono; Boss CL50 compressor/limiter, boxed, £100 ono, AKG414 UBLS microphone, classic, mint condition, £450. ☎ 0378 641400/01293 823631 (Surrey).

SONY TCD D8 DAT recorder/player, best portable DAT, boxed, manuals, £350, no offers. ☎ 01753 546413 (Berkshire).

SOUNDCRAFT GHOST, 72 channel mix-down, 17 aux-sends, 4-band EQ with 2 fully parametric mids, meter bridge, mint condition, 6 months old, boxed, manuals, receipt, £3000. ☎ 0181 778 0008.

SOUNDTRACS SOLO MIDI 32, 72 inputs, 6 auxs, 4-band EQ, MIDI muting on all inputs, bought new for £2300; offers please; Midimation 1U rackmount unit, 16-channels of fader and mute automation, suitable for any desk, works from Cubase, Notator and similar, £350 ono. ☎ 01633 613342 (Newport).

SOUNDTRACS TOPAZ MINI 12:2 mixing desk, perfect condition, never used, boxed, manual, £175 ono. ☎ 01253 874107 (Lancs).

SPIRIT 16-CHANNEL live mixer, £600;

Yamha P1500 studio amp, £250; Behringer Denoiser, £125; Trace Elliot 4x12 cab, £260; Gibson Epiphone with 3 Hotrail pick-ups, case, £150. ☎ 0114 288 2236 (Sheffield).

STUDIOMASTER PRO LINE GOLD, 16:8:16:2 mixer with MIDI muting, recently used to record master tapes for a CD, £650 ono. ☎ 01793 539993 (Swindon).

TANNOY LITTLE RED studio monitor, one only, good working order, offers. ☎ Alec 0181 333 0265.

TASCAM 424Mk2 4-track tape recorder, 3-band sweepable EQ, 2 auxlary sends, 4-track simultaneous recording, new condition, £300. ☎ Oliver 01481 63113.

TASCAM 424Mk2, 4-track recorder, boxed, hardly used, in excellent condition, £325 ono. ☎ 0121 705 1286 (West Midlands).

TASCAM 464 4-track with built-in 12-channel mixer, tape sync unit, light home use, boxed, manual, £350; Korg 01/W librarian Editor, £40. ☎ 01933 270930 (Nottingham).

TASCAM 488 8-track portastudio, home use, excellent sound and condition, £475 ono. ☎ 01275 817026 (Bristol).

TASCAM 488, 8-track cassette portastudio, hardly used, excellent condition, in box with instructions, £450. ☎ Stuart 0161 366 5299 (Manchester).

TASCAM 488 MK2 8-track portastudio, XLR inputs, 2 aux, dbx, 3-band EQ, sweepable mids, mint condition, manual, £450. ☎ 0121 705 7893.

TASCAM 80:8 8-track half-inch recorder, dbx noise reduction, remote, cables, 5 mics, stands, connection box, tapes, excellent condition, £795 ono. ☎ 0976 533102/01384 351033 (Stourbridge).

TASCAM MM1 mixer, 20:2, MIDI mutes, £220; Ultra Metal guitar effects pedal, £20. ☎ Carl 01507 606956 (Lincs).

TASCAM M2524, 24-channel 8-buss mixer with MIDI muting, parametric EQ, £795; Boss SE50 effects, £150; Behringer Ultrafex, £50; all in good condition, can deliver. ☎ Paul 0121 359 1232/0958 287231 (Birmingham).

TASCAM 112 professional cassette recorders, 10 of them for £250 each; or £2000 for the lot, no offers. ☎ John 01449 673695 (Suffolk).

TASCAM TSR8 professional half-inch, 8-track recorder and remote control, superb condition, only £900; many other home studio items. ☎ 01579 320024 (Cornwall).

TASCAM 2-CHANNEL dbx noise-reduction unit, rack mounting, £50, 10 reels of half-inch Ampex tape, £50; Fostex 280, £300 ☎ Steve 01904 470854 (York).

TEAC A3440 reel-to-reel 4-track machine, mint condition, tapes, £149. ☎ 01246 230543 (Derbyshire).

TEAC A7300 open reel 2-track stereo master recorder, 15ips with 10-inch reel capacity, a professional machine, £400; Marshall 9004 stereo rack-mount guitar preamp, boxed with PSU, £95; Roland Pro E intelligent arranger, similar pro-spec to D110 (128 voices) but with 3 octaves, automatic accompaniment/sound effects, manual £220; Yamaha FB01G MIDI sound generator/expander, good working order, manual £95; Watkins (WEM) Copicat tape echo machine, good working order and in excellent condition for age, £95. ☎ 01329 238522 (Fareham).

TUBETECH MP1A mic preamp, £900, C1 1A compressor, £850; PE 1B EQ, £700; all studio use only. ☎ Gary 01869 810956 (Oxfordshire).

WEM COPYCAT original, £90; flightcase rack/sleeve, rackmount top and front, 16U, castors, £180; Speaker cab, £280. ☎ 01277 630283 (Billericay).

YAMAHA EMT10 MIDI sound module, pianos, guitars, vocals, immaculate, £75; Beyer DT100 headphones, £65; MXR vintage stereo compressor, immaculate, owned from new, £275. ☎ 01482 448767.

YAMAHA M15/24 professional 13 buss mixer, 48 inputs, a host of pro features, £850 ono; Yamaha FX770 guitar rack-effects, £250 ono ☎ Dave 01785 662896 (Stafford).

TOA MRT8 8-track cassette multitrack recorder, 8-track simultaneous recording, HX Pro panning effects-loop, £400 ono. ☎ Colin 01473 464555 (Suffolk).

YAMAHA MT120 multitracker, £85; acoustic guitar amplifier, £70; Yamaha Rex 50 guitar effects, £75; digital orchestrator plus MIDI/audio sequencer, £75; Maxisound 64 soundcard, £75; SW60XG, £75. ☎ 01827 703562 (Staffs).

YAMAHA MT120 4-track, high-speed play and record, dbx stereo 5-band EQ, effects loop, perfect condition, £260. ☎ 0850 386093/01564 778009.

YAMAHA MX200, 24:2 desk, as new, £365; Novation Bass Rack, £185; Roland R8, £285; Ensoniq SD1 synth/sequencer, £495; all mint with manuals, cash needed. ☎ 0370 661511/01253 872937 (Lancashire).

YAMAHA PROMIX £700; Korg 1212 PC card, £400; Ensoniq QSR, £300; Korg NS5R module, £330; Philips DCC, £120; Roland D5 keyboard, £160; D10, £175; D110, £110. ☎

01252 377638 (Surrey).

YAMAHA PROMIX 01, 16-channel megamixer, excellent EQ, motorised faders, digital output, versatile effects, rack ears, home use only, boxed, immaculate, £675. ☎ John 01392 219323 (Devon).

YAMAHA PROMIX 1, £640; Ensiniq SQ1+, £225; Roland JX3P, PG200, £289; PG800, £180; MVS1, £190; W30 SCSI, £395; MS10, £169; Kawai K4R, £175; TR606, £90, will post. ☎ 01261 815707.

YAMAHA RM800 mixing desk, 24:8:2, 24-track, 48 channels, 4 aux sends, 3-band EQ, mid sweep, home use only, mint, £800 ono. ☎ Rob 0116 251 0405/276 1242 (Leicester).

ROLAND MC4B microcomposer with Roland OP8M CV interface, allows programming up to 8 simultaneous CV to gate tracks, mint, never used, £350 ono. ☎ Chas 01932 348051 (Surrey).

ROLAND MC300 8-track MIDI sequencer with built-in disk drive, manuals, excellent condition. ☎ 0171 372 3772

ROLAND MC303, in very good condition, can connect to MIDI via second computer port, manuals, £300 ono. ☎ Derik 01229 582822 (Cumbria).

ROLAND MC303 groove box for sale, latest ROM version, mint, manuals, £250; leave a message on voice mail. ☎ 0151 678 0607 (Wirral)

ROLAND MC303 drumkits, sounds, sequencer, brilliant machine, £365, Yamaha SU10 portable sampler, filters, sampling time of 1 minute 10 seconds, £160, both boxed, manuals, great for dance music, will accept £475 for both ☎ 01274 591329.

ROLAND MC505, six weeks old, £750; MC303, £300; MS1 digital sampler with 4Mb upgrade, £200; will part-exchange, Akai S2000 or Roland TB303. ☎ Paul 0411 333550 (Portsmouth).

ROLAND MT100, 5-track digital sequencer and sound module, disk drive, 500+ LA sounds, 9 part multitimbral, 32 note polyphonic, manual PSU, flightcase, mint, £150, or swap for Zoom 1204 or SU10. ☎ 01639 768103 (South Wales).

YAMAHA QY70, as new, still boxed, many extras. ☎ 0171 252 3158.

AKAI S20 sampler, fully expanded, sample disks, £300. ☎ 01922 683396 (Walsall).

AKAI S900, manual, 8 drum-trigger cards, upgrade software, 8 outs, excellent first sampler, warm sound ☎ Martyn 0181 675 0120.

AKAI S900 sampler, 10 outputs, library, ideal for drums and percussion, great condition, smart 12-bit resolution, manual, will accept £299. ☎ 0161 442 8959 (Southport)

AKAI S900 sampler, 8 outputs, TR909 sample disks, £295; Korg SDD3300 triple sampling digital delay, boxed, £250; Yamaha QY10, £50. ☎ 01268 525347.

AKAI S1000 v4.4, 26Mb, SCSI, 10 outputs, large library, never gigged, excellent condition, boxed, manuals, £975, Lexicon LXP1 and LXP5 in a 1U rack, £500, Ensoniq ESQ1 8-track sequencer, £200. ☎ Jamie 0966 302290 (London)

AKAI S1000/S1000PB, 2Mb each, library, £1250, Korg X3, £520, Roland U20, £310, Seck 24:2 desk, £415, Boss SE50, £210; Simmons SDE expander module, £55. ☎ 0700 900 1153/0411 744154 (London).

AKAI S1000PB playback sampler, large Akai library, good condition, £380; Quasimidi Rave 'O' Lution with input expansion board, £475. ☎ 01442 393892 (Herts).

AKAI S1100, 10Mb, dig I/O, effects, SCSI with zip cable, boxed, original manual, 40 Akai library disks, great condition, £780. ☎ Oisin 0171 241 1093

AKAI S2000 stereo sampler, 6Mb, 8 output board, boxed, editing software, perfect condition, will accept £600 but open to offers. ☎ Will 0115 911 5754 (Nottingham).

AKAI S2000, 10Mb, Quad effects board, zip, mint, £800 ono; Roland U20, mint, card, £350; ES132 sampler wanted and Roland U220. ☎ Rob 0116 276 1242/251 0405 (Leicester).

AKAI S2000, 48Mb, expandable to 64Mb, CD-ROM drive and 3 CD-ROMs, offers please. ☎ 01685 811593 (Glamorgan)

AKAI S2000 stereo sampler, 32Mb version, 2.0 software, boxed with manual, in very good condition, £700; Yamaha TG33 vector synth, £150. ☎ Simon 0161 8606139 (Manchester).

AKAI S2800 sampler, excellent condition, boxed with manuals, latest OS/OS 4, outputs, EFX, £650 ono; Zoom 9030 effects, 47 different effects, 8 simultaneous, with manuals and PSU, half-rack size, £150. ☎ Dale 0161 872 9064/0956 424619
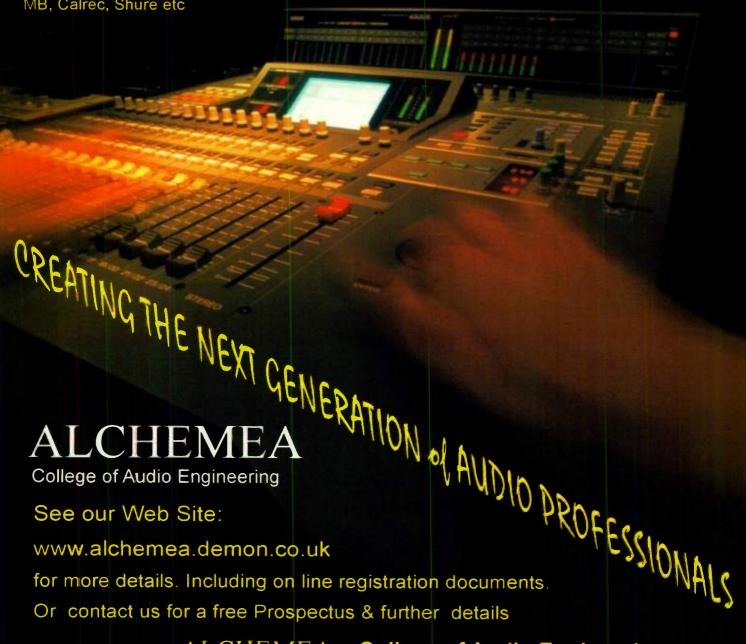
AKAI S2800 sampler with effects and 10Mb of memory, £630 ono, Philips colour Atari monitor, £40, Art Alpha SE studio effects, £90, all in excellent condition. ☎ 01483 561430.

AKAI S3000XL, 16Mb, will swap for MPC2000 with an expansion ☎ Simon 0181 599 4668 (Essex).

AKAI S3000XL, 18Mb, zip, £1100, Akai MPC 2000 sequencer/sampler, 18Mb, £1000, Roland JV2080, £1100; Boss SPX700, £250, Spirit Folio SX, £350 ☎ Kenny 01344 778899 (Berks).

AKAI SXL3000 sampler, hardly used, £816 effects processor expanded to 16Mb, £850 ono ☎ 01706 211131 (Lancs).

EMU ESI4000 sampler, turbo effects and digital I/O installed, box and boxed as new, £1000 ☎ Peter 0171 435 3531.

CASIO FZ1 16-bit keyboard sampler, 8 outputs, resonant filters, very good condition, disks, manual, £280 ☎ 0468 173939 (Chester).

CASIO FZ1 sampler, 61 keys, velocity and aftertouch, 8 separate outputs, £250 ono ☎ Rob 01782 714371 (Staffs).

EMU ESI 32, 8Mb sample, Zoom 1204 MIDI effects module, Roland Juno 1 keyboard, Cubase VST for PC, all complete, mint, £1200 ☎ 01226 751091 (Barnsley).

EMU ESI4000, 17 filters, time-stretch, doppler effects, SCSI, 32Mb, 64-note polyphony, can expand to 128Mb, £799 ☎ Gareth 0831 569181 (Leeds).

EMU EMAX II keyboard, 4Mb memory, in very good condition, £450 ono, Kawai RV4, 4 independent stereo-effects, £280 ono, Simmons SMP 8.2, MIDI controlled mixer, £120 ono ☎ 01732 354665 (Kent).

EMU EMAX II sampler, large library, original Emu disks, SCSI interface, 8 outs, original manual, £400 ono ☎ 01932 782929.

EMU EMAX II stereo sampling keyboard, home use only, memory expanded to 8Mb, including manual and disks, £600 ono ☎ 0191 232 4175 (Tyne & Wear).

ENSONIQ EPS16+ rackmount sampler, library,mint condition, manuals, £425 for quick sale ☎ Paul 01285 642004 (Glos).

KORG DSS1 synth/sampler, mint, book, disks, £250 ono ☎ 01444 250482 (Sussex).

ROLAND DJ70 stereo sampler, £350 ono ☎ 01275 462762 (Nick).

ROLAND DJ70 sampling keyboard, fully upgraded, lots of sounds, £550, Korg 01/W workstation with flightcase, £550, Korg Wavestation, flightcase, £550 ☎ 0171 358 1903.

ROLAND DJ70, 4Mb, £350, Akai S900, £250, both in good condition and with manuals, Atari Stacy laptop, 2Mb, 40Mb hard drive, music software, £300 ☎ Ian 01389 763147 (Scotland).

ROLAND MS1 16-bit stereo sampler with 5Mb flash RAM-card, manuals, boxed, excellent condition, £250, Mackie 1402VLZ 14-channel mixer, boxed with manuals, excellent condition £300 ☎ 01603 461450 (Norwich).

ROLAND SAMPLER monitor, 14-inch display, suitable for S330, S550, S760 and S770, complete with special R&B cable, can despatch if required, £85 ☎ 0181 763 8991.

ROLAND S50 sampling keyboard, 4 outputs, on-screen display, manual, £300 or £350 with Philips RGB monitor ☎ 01638 602955 (Suffolk).

ROLAND S220 rack sampler, 4 outputs, £95; Emax 1 rack sampler, £150, Alesis Quadraverb+, £190, 100W guitar/keyboard combo, £95, Yamaha QX5 sequencer, £40. ☎ 01342 323094.

ROLAND S760, 18Mb, CD-ROM, zip, boxed as new, £950; Korg M1R, boxed, £349; Proteus,1, boxed, £229, Atari STE, 4Mb, monitor, software, £249, Emu DK4. ☎ Geoff 01723 351239.

ROLAND S760 sampler, 10Mb RAM, OP-760-1P digital I/O board and Syquest EZ135 drive, box and manuals, immaculate condition, home use only, £1100 ono. ☎ 01222 222339 (Cardiff).

SEQUENTIAL PROPHET 2000 sampler, need new disk drive, £300, PC 486DX66, Cubase v1, Cookit Pro, £250; Roland SH101 mod grip, grey, £50; Cubasis Audio, £90; D50 card 00-03, £30 each; ESQ 1, expanded sequencer, cartridge, manual, £375; CZ1, £200; Bit99, £175. ☎ 01902 744293 (Wolverhampton).

SEQUENTIAL PROPHET 2000 sampler keyboard, 61 weighted velocity-keys, arpeggiator stack model, extensive MIDI specification, disks, manual, in very good condition, £330. ☎ John 01389 751843 (Dumbartonshire).

YAMAHA A3000, 32Mb, £900, Quasimidi Quasar, £400; Emu Orbit v1, £400, Yamaha TG55, £150; Yamaha TG33, £130. ☎ 01274 827830 (Bradford).

YAMAHA SU10 mini sampler, PSU, sample CDs, immaculate, boxed, great sound, £195 ☎ 0151 630 2098 (Merseyside).

DRUM MACHINES

ALESIS SR16 drum machine, boxed, hardly used, as new, £150 ☎ Paul (evenings) 01280 706899 (Northants).

BOSS DR550 great condition, has TR808, TR909 kits, easy to use, great for recording backing tracks, £120 ono ☎ 01733 577238 (Peterborough).

BOSS DR660, 39 kits, 255 editable sounds including 808/909, 4 outputs, reverb and chorus built-in, in very good condition with power supply, £180 ☎ 01933 678608.

ROLAND R8 MkII, immaculate condition, £400, must be collected ☎ Ian 01322 409106 (Kent).

ROLAND TR626 drum machine, great collection of sounds, intuitive Roland programming, 8 individual outs, MIDI, will run on batteries, with PSU and manual, good condition, £150 ☎ 01354 695239.

ROLAND TR707 and TR727 rhythm composers, full working order, very good condition, home studio use only, box and manual, £110 each or £200 for the pair. ☎ 01952 260064 (Wakefield).

ROLAND TR808, needs attention, £200, Yamaha RX5 with manual, no RAMs or ROMs, £200 ono ☎ Kris 0151 523 8110.

ROLAND TR909, £850, Roland TB303 with Zoom 1201 effects box and sync box, £750; Roland MC303, £350; Boss DR660, £180. ☎ 01579 343174 (Cornwall).

ROLAND TR808, fitted with MIDI in, out and thru, very powerful, perfect condition, £500; DOD 15x2 graphic EQ, £120. ☎ Nick 0181 361 3544.

ROLAND TR909, good condition, manual. ☎ James 01978 362910.

SEQUENTIAL CIRCUITS Drumtrax, £250 ono, the person from Hove who wanted a copy of a manual - I lost your address. ☎ Jon 0113 293 6896 (Leeds).

SIMMONS SDS2000 drum brain, manual, £90, Cheetah MK5V master keyboard, £60 Premier olympic drum kit including Paiste 18-inch ride, £50 ☎ 01536 501926.

SIMMONS SDX, intelligent pads, 8Mb sample time, 16 outs, SCSI, MIDI sequencer, in very good condition, manual, £1500; Roland JD800 keyboard, immaculate, manuals, £850, Atari 1040 STE, monitor, Cubase, £160, Soundcraft Spirit live mixing desk, 16:3, £500 ☎ 0181 748 9322/0171 313 0081.

YAMAHA RY20, great sounding drum and bass patterns, as new, only 3 months old, boxed with PSU and manual, £220, Quick Loc QL646 heavy-duty double X stand, brand new, cost £48, will accept £30 ☎ Gordon 0181 985 7422.

YAMAHA RY30 drum machine, Atari music software, Micro Deal sampler, Intel 486SX, Intel overdrive DX20 DP66 simms ☎ Martin 0181 577 6974.

PERSONNEL

DRUMMER REQUIRED to join a band performing music from our latest album release, must be local (near Preston) and fantastic, recording of next album to follow soon ☎ Neil 01772 816508 (Lancs).

ESTABLISHED PRODUCER with chart track-record, offers unique opportunity to acts with commercial potential, fantastically cheap rates, this is a one-off special-offer, dance acts especially welcome ☎ 01379 853982 (Norfolk).

EXPERIENCED FEMALE SINGER from Austria, seeks band or singing group, not too far from Ledbury, Herefordshire ☎ Annibe 01531 633085.

HAVE YOU WRITTEN A POTENTIAL hit song? Beautiful young female artist with contract, seeks songwriters to write/co-write material for recording ☎ Kirsten 01277 218842 (Brentwood).

OLD, same age as Utley, keyboards/Cubase, inventive, different, looking for collaborator (preferably female) with interesting voice and creative input, feet on the ground please ☎ 04325 270082 (East Devon).

PROMOTER/MANAGER required urgently, young female singer with 3 CDs, into drum & bass/dance, looking to gig the London scene, only professionals need apply. ☎ 0181 257 8713.

STUDIO EXPERIENCE sought by keen 28-year old, currently undertaking a recognised course Herts/Beds/North London. ☎ Clem 01582 423791 (Beds).

TRAINEE ENGINEER seeking experience in voice recording and editing wanted, knowledge of computer-based systems and editing desirable, must be presentable with a sense of humour, call Voice Caster. ☎ Wendy or Steve 0171 624 1900 (NW1).

VOCALIST WANTED, Ann Wilson/Stevie Nicks for original acoustic and electric project, well known guitarist involved, songwriting skills a bonus, must have a great voice, big production and gigs. ☎ 01604 637509 (Northants).

COMPUTERS/SOFTWARE

APPLE CD300E CD-ROM, external SCSI drive, ideal for sampler or computer with manuals, £85, Akai CD-ROM electro shock double, £25 ☎ 01933 678608 (Northants).

APPLE MAC LCII, 10Mb RAM, Cubase VST, Midiman 2 in, 6 out interface, 14-inch colour monitor, various software, fax, modem, keyboard, mouse, £249 ☎ Alan 01494 677106 (Bucks).

APPLE MAC 7200/90, 1.2Gb hard drive, 32Mb RAM, monitor, keyboard, mouse, £695 ono, Roland XP50, £595a, Korg 01W/FD, £595. ☎ Steve 01429 222517/295838 (Hartlepool).

APPLE MAC 7600/64, 1Gb, monitor, keyboard, Cubase VST, lots of plug-ins, Time Bandit, Recycle, Protools 4, may split, £350, Volt Pro Studio 35Hz, monitor, speakers, £400, Amcron D60 stereo studio amplifier, £110; Roland PMA5, £300 ☎ Mark (after 2pm) 01924 216047 (Wakefield).

APPLE POWERBOOK 1400CS, bundled with Emagic Logic v3, Steinberg Recycle and Rebirth, MOTU MIDI interface, all still boxed, perfect condition £1350. ☎ Andy 01442 878370 (Herts).

APPLE POWERMAC S200, 16Mb RAM, 500 hard disk drive, modem, Cubasis, MIDI filters, Word, Pagemill, Photoshop, Internet Explorer, perfect condition, £695 ☎ 0973 632405 (Staffs).

APPLE POWERMAC, 200mHz, 64Mb RAM, 2Gb, Cubase VST, Roland MV30 music station, all leads, will deliver, £3600 ono. ☎ 01970 627700/01545 570601.

ATARI C-LAB editor for all DX keyboards and modules, £40, Atari box of spares, drives, cables, joystick, £25. ☎ 01865 776587 (Oxford).

ATARI FALCON, 14Mb, 128Mb internal hard drive, 32mHz accelerator, extras, 14-inch SVGA monitor, £440, Cubase Audio v2.06, £380, CD recorder Pro 2, £150. ☎ 01392 251552 (Devon).

ATARI ITEMS, Cuetrax, Styletrax, studio modules, £85, Steinberg Key-expander, £45, Cubeat 2, £10, Replay 16 sampler, £40, Midiman minimixer, noiseless 4 by 2, stereo 8 by 1 mono, steel case, £40. ☎ 0117 965 9211 (Bristol).

ATARI MEGA ST2, SM124 monitor, 20Mb hardrive, Cubase, other software, includes box and manuals, in very good condition, £280, will swap for Quasimidi Quasar ☎ 01462 678685 (Herts).

ATARI MIDI PORT EXPANDER, gives 32 MIDI channels, comes with Cubase drivers, £20 ☎ 0141 638 6687 (Glasgow).

ATARI STE, 4Mb, mouse, manual, Atari productivity software ☎ Dave 0181 656 0434.

ATARI STE, 4Mb, Cubase 3, new Lynwood Pro monitor, excellent condition, £190, Phonic MM122 12.2 mixer, £130, Kawai K1r, 1U rack, lovely, 3 cards, £150 ☎ 0181 402 1108.

ATARI STE with SM144 monitor, 4Mb, Midex+, £250, Roland D110 module, £150, Proformance piano module, £150; boxed, manuals, Steinberg M1/D110 editors, £50 each ☎ John 01530 243865 (Leicester).

ATARI STFM with sequencer, £50, Behringer Studio amplifier, £10, Steinberg Pro 24, £25, Laney Bass amplifier, £60; Yamaha YST M10 powered monitors, £30, Electrovoice EVMC100 mic, new, £50 ☎ 01536 501926 (Northants).

ATARI STFM, monitor, printer, software, £170 ☎ 01827 53220 (Staffs).

ATARI 32-CHANNEL MIDI port adaptor, converts modem port to add 16 channels to Cubase, £15 ☎ 01375 401964/0411 592283.

ATARI 520 STE, needs attention, 2Mb, SM124 monitor, 2nd floppy drive, low/medium resolution monitor, loads of music software, £85 for the lot. ☎ Reds 0498 728886 (Oxford).

ATARI 1040 with monitor, original C Lab Notator sequencer program with dongle, £200 ono ☎ 0151 513 1106 (Merseyside).

ATARI 1040 ST, SM124 monitor, £225 ono; Kawai MX8R 8.2 rack mixer, ideal for small set-up or as a sub-mixer, £150 ono ☎ Alon 0181 525 1918.

ATARI 1040 ST computer with SM124 monitor and a copy of Cubase v3, plus a 90-minute tutorial video, £120. ☎ 0114 248 4440 (Sheffield).

ATARI 1040 STE, 14-inch monitor, swivel base, high-res monitor, fully upgraded to 4Mb RAM, Cubase v3.24 and tutorial, good condition, possible delivery, £230. ☎ 0181 656 0291.

ATARI 1040 STE, high-res monitor, Cubase V2, £200, Cubasis CD-ROM for PC/Mac

compatibles, £60. ☎ Steve 01553 674076 (Norfolk).

ATARI 1040 STE, 2Mb, Atari monitor, £160; Cubase Audio for Falcon, dongle, manual, £210, ADAT interface for Falcon, £235, 16Mb memory for Falcon £50; Fostex R8, MTC1, £575, Sony DTC750 DAT recorder, £350. ☎ 01702 521570.

ATARI 1040 STE, 4Mb, SM125 high-res monitor, 20Mb hard drive, Cubase, MIDI leads, £300 ono, Zero X for ST with dongle, ST Replay 16, boxed as new, £80 each, no offers ☎ Dan 01737 219083.

ATARI 1040 STE, 4MB monitor, mouse, £160. ☎ 01562 720351 (West Midlands).

ATARI 1040 STFM, 1Mb RAM, high-res monitor, replacement mouse, £100 ☎ Aaron 01904 416212 (York).

CAKEWALK v6.01 home studio, as new, boxed, manuals, CD-ROM for Windows 95, cost £100, will sell for £65 ☎ Lawry 0151 922 9176 (Merseyside).

CUBASE PC on CD-ROM, pre VST, 8-track hard disk recording, version with dongle, manuals, Udo Weyer's guide, £150, Philip Rees MIDI Merge, £30. ☎ Tony 01427 614135 (Lincolnshire).

CUBASE SCORE v3 for Windows, manuals, dongle, £150. ☎ 01642 658841 (Cleveland).

CUBASE VST, PC Waves, Audiotrack, Winman 2 by 2 MIDI interface, £250, Turtle Beach Fiji with digital, £220, Shure SM58, £50, 10U flightcase, £75 ☎ Stuart 0115 904 3526 (Nottingham).

CUBASE VST for PC with dongle and manual, still in box, yours for £195, Compaq Pentium with sound card, £450 ☎ 01203 667568 (Coventry).

DIGIDESIGN PROTOOLS 4.1, 16-track PCI, core-system and masterlist CD, boxed as new, £3000, Roland Portable MIDI sequencer, PMA5, £300 ono ☎ 0171 831 8155.

DIGITAL AUDIO LABS card D PC soundcard, industry standard, excellent quality, you know the spec, also the I/O card D, which adds S/PDIF, £450 for the pair ☎ 0113 2406921 (Leeds).

PANASONIC CD-ROM DRIVE for Akai/Emu samplers with SCSI cable, brand new, boxed, cost £180, never used hence £115 ono. ☎ 0151 339 4404 (Wirral).

POWER MAC 604e, 200mHz, 2Gb hard drive, 24X CD, 512 L2, 64Mb RAM, Cubase VST Score 3.5.2 with 25 of the best plug-ins for Mac, Rebirth, Recycle, Encore, all original, best offers ☎ Dan 07771 728110 (Norfolk).

VALE 486 DX2 50, 8Mb, CD-ROM, 16-bit soundcard, 1Mb cache, Cubasis, Win 3.1, A3 dot-matrix printer, no monitor, £180 ono ☎ 01900 828312 (Cumbria).

YAMAHA CBX D5 with Logic Audio extension software, £650; Roland E660 digital parametric EQ, £375; Quasimidi Technox, £400; Peavey Spectrum Bass, £185, all in very good condition with manuals ☎ Adam 0113 216 5177 (Leeds).

YAMAHA CBX D5 hard disk recorder, excellent condition, £475; Tascam DA20 DAT machine, box, hardly used, £450. ☎ Rob 01844 217833 (Oxon).

MISCELLANEOUS

CARLSBRO GRX7, 400W mixer amp with 2 Yamaha PA speaker-cabs, in very good condition, £800. ☎ Mark 01452 780131 (Gloucs).

CURTIS CHIPS:, 5x CEM 3340, 5x 3310, 3x 3320, 3x 3330, 2x SSM2044, best offers, swaps/part-exchanges possible for decent sample or interesting MIDI/analogue gear ☎ Chris 01235 534152 (Oxon).

FENDER PRO ROC70, 2-weeks old, £280 ☎ 01225 460520 (Bath).

FORD CARGO MOBILE studio, 7.5 ton, 6 cylinder diesel (non HGV), 20 by 8 by 8-foot control room with spotlights, carpet, acoustic insulation, alarm sleeper cab, tail lift, MOT, serviced, in very good condition, £6991, UM92S valve mic, sus-mount, case, £795, Joe Meek classic stereo compressor v1.05, £749. ☎ 0181 875 9712.

GIGAPACK DOUBLE CD-ROM, for Roland samplers, covers most library area and incorporates classical chiors, was £260 new, sell for £60. ☎ 01132 783690 (Leeds).

LES PAUL STANDARD, wine red, 1980, good all-round condition, case, £850 ☎ 01323 736017/07771 628473.

MARSHALL ANNIVERSARY half stack, £725; Peavey VSX cross-over, £150; Peavey Q215F twin 15-band graphic, £175, Peavey CS800 amp, £325 ☎ Steve 01904 470854 (York).

MARSHALL REVERB 12W combo amp, model number 5205, absolutely mint condition, hardly used, cover, great sound £70 ono, Drum Practice Kit, comprises a set of 5 rubber covered metal pads with stand, very robust design, Paiste 101 16-inch crash cymbal with stand, zildjian 18-inch crash cymbal, stand, kick drum pedal, good

condition, £95. ☎ 01733 391635 (Peterborough).

MERCEDES 508 diesel panel-van, MOT, taxed, can seat six, A-reg, low mileage, alarm, load partition, tow bar, good gig van, £1350 ono ☎ 01206 323221 (Colchester).

MOBILE STUDIO, V8 Perkins BBC outside broadcasting truck, fully soundproofed, solid aluminium body, no tax/MOT, engine needs attention, currently a live-in vehicle, £2000 ono ☎ 0374 257716 (Milton Keynes).

MULTICORE EDAC, in flightcase, 30 core 'A' cable, 50 meters, plus 19-inch core, 'B' cable with stage box, fully working, would suit a large band set-up, £500. ☎ Ian 01493 384407 (Norfolk).

OPTICAL DRIVE, 2.0Gb, 6 blank disks, mint condition, £450. ☎ 0171 435 2326.

RACK-MOUNTABLE portable equipment case, as new, 19-inch by 19-inch, only £65 ono, Printer Star colour LC 100, £30. ☎ 01722 744196 (Wiltshire).

REGA TURNTABLE, Planar 2, Pioneer earphones, excellent condition, £130. ☎ 01300 348226 (Dorset).

STEREO PA POWER amplifier, 100W per channel, 19-inch rack-mountable, VU meters, protector circuitry, excellent condition, 3 months old, £110 ono. ☎ 01539 730335 (Cumbria).

UNIQUE ONE BEDROOM LIVE & WORK loft apartment, 1000 square foot with a further 250 square-foot basement storage, double height throughout, exposed brickwork and timber, oak flooring, SKK light-fittings, Clyde radiators, sound proofed, air-conditioning bedroom and dressing room, Studio and vocal room, mezzanine and storage areas, alarm system, lots of daylight, private road owned by freeholders, great views of St Lukes, short stroll from Barbican/Moorgate/Old Street, £250,000 including share of freehold ☎ 0171 253 4730/0850 930096.

WANTED

AKAI REMIX 16 wanted, will pay up to £300 ☎ William 01226 246871 (Barnsley).

AKAI S1000 with SCSI card wanted ☎ 0171 837 7733.

ANALOGUE KEYBOARDS WANTED, Yamaha CS30, Crumar Spirit, Korg MS20 or SH05 ☎ Pete ☎ 01286 872653 (North Wales).

ATARI SOFTWARE, Steinberg studio moduleand SP edit for Peavey sampler, cash waiting. ☎ 0171 251 1920.

CALREC Mk4 Soundfield microphone wanted, in top condition, please call during office hours. ☎ 01494 787416 (Bucks).

DIGIDESIGN 888 interface for Protools, £500. ☎ Martyn 0181 906 2057.

ELECTRO HARMONIX guitar synth, in any condition, Mutron, A/DA products ☎ (evenings) 0121 420 3295 (Birmingham).

ELKA DMP18 MIDI bass pedals wanted, any condition ☎ Mark 0979 724059 (Jersey).

EMU EMAX 2, minimum 4Mb plus library, operating system, swap for Akai S950 fully upgraded and cash difference ☎ Drew 0181 505 5298.

EMU ESi 4000, will pay £800. ☎ Mali (pager) 01426 159525 (Suffolk).

FAULTY EMS, Synton Syrinx ☎ 0121 420 3295 (Birmingham).

FOR HOME STUDIO, AN1X, JP8000, S3000XL, E6400, TB303, TR909, Drumstation, Trinity, A3000, Studioquad PCM80, JD800, Orbit, ADAT, DAT, NS10M, AB2, MAckie or similar, cash waiting ☎ 01252 677059 (Hants).

KORG M1 OR X3, must be in good condition and should include all manuals, Roland TB303 or TR909 also wanted ☎ 01485 570568 (Norfolk).

MOUSE for Roland S760 sampler ☎ Steve 01722 336395 (Sailsbury).

MORPHEUS EDITOR, Librarian, sounds wanted, PC or Atari, RAM cards, ROM cards ☎ Steve (evenings or weekends) 01270 820393 (Stoke).

PC EDITOR/LIBRARIAN wanted for a Roland D110, user friendly graphical interface (Atari Synthworks style), will pay cash or swap. ☎ 01752 562343 (Devon).

PPG WAVE 2.3 with or without Waterm 13, cash waiting, Pro s, Rev 2 MIDI, £875 ono, complete Roland DM80 system faders, mixer remote, offers ☎ 0836 667799 (Northants).

ROLAND JV880 with vintage expansion board, Roland JV1080 with vintage expansion board, JD990 also wanted ☎ 0181 449 9698 (Barnet).

ROLAND R8 electronic card, Roland D70 ☎ Tim 01788 815171 (Rugby).

ROLAND U110 Rom cards, any brass ☎ Mick (after 6pm) 01203 306130 (Coventry).

SOUND CARDS for Roland D110, all types considered, cash waiting ☎ Alistair 01763 271227 (Herts).

World Radio History

World Radio History

World Radio History

HOW TO SUCCEED IN THE MUSIC BUSINESS

Alan Dann and John Underwood
Completely updated and revised from the original edition, this has become the classic guide to avoiding the pitfalls that can beset a newcomer in the music industry. The book contains sections on songwriting, recording and artist management, as well as sample contracts and a general guide to the kind of terms offered by publishers, record labels, managers and agents.

CODE B360   **£8.99**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

MIDI INFORMATION

MIDI FOR THE TECHNOPHOBE

by Paul White
In this book the absolute beginner is introduced to the concept of MIDI by way of analogy with familiar technology. Myths relating to the dehumanisation is introduced in a straightforward, logical order.
In just a few hours, with no confusing jargon, you'll find out how MIDI works, what hardware is needed to build your own MIDI system and how to use that system to create, record and edit your own music.

CODE B355   **£11.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

MUSIC TECHNOLOGY REFERENCE BOOK

by Peter Buick & Vic Lennard
Written by Peter Buick and Vic Lennard (author of SOS's Atari Notes column), this book offers information at your fingertips and is an indispensable reference source for the musician or sound professional in the recording, broadcast, live, video, computer, multimedia, post production and theatre industries. It includes comprehensive MIDI spec, General MIDI, MTC, MSC, and SysEx information. It also includes charts, check lists, useful hints, tips, and ideas, plus a glossary, list of contacts and a comprehensive indexed. It is also organised into convenient sections. 150 Pages.

CODE B305   **£12.95**
Postage: UK £1.95, Europe £5.50, R.O.W. £8.50

WHAT'S MIDI

by Jon F Eiche
This slim 82-page book introduces MIDI in very simple terms. If you're a total beginner, then this could be an essential purchase.

CODE B347   **£3.95**
Postage: UK £1.95, Europe £5.50, R.O.W. £8.50

THE MIDI COMPANION

by Jeffrey Rona
A recently expanded edition of a popular book. The new edition contains a comprehensive rundown of all areas of MIDI, with helpful explanations of System Exclusive, using computers, troubleshooting and new explanations of the General MIDI standard.

CODE B348   **£7.95**
Postage: UK £2.50, Europe £5.95, R.O.W. £9.50

PLAYING TECHNIQUES

MUSIC IN SEQUENCE

by William Lloyd & Paul Terry
Still one of our best-selling books. Easy to follow, practical guide to making real use of your sequencer, written by people who really understand music and how to create it using today's technology. This brilliant, unique book carries the SOS seal of approval.

CODE B155   **£12.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

1000 KEYBOARD TIPS

by J Drekster & D Harle
The hi-tech musician's guide to the basics of music, scales, chords and harmonies. 256 pages covering chord relationships, improvisation, suggested drum machine patterns, plus comprehensive chord table. Too good to miss!

CODE B113   **£13.95**
Postage: UK £3.50, Europe £6.95, R.O.W. £9.50

THE MIDI FILES

By Rob Young
This book comes from Prentice Hall and aims to help you get the most out of your MIDI system, by providing loads of tips on programming realistic instrument and rhythm parts, using sequencers to create advanced effects. Basic MIDI concepts are covered, along with advice on setting up a MIDI-based home studio. More advanced discussions cover specific types of MIDI messages and what you can achieve with them. All the techniques in the book are illustrated on a supplied floppy disk.

CODE B334   **£19.95**
Postage: UK £2.50, Europe £6.95, R.O.W. £9.50

MAKING THE MOST OF MIDI

By Paul Overaa
This is a MIDI book unlike any other. It explains the fundamentals of MIDI and MIDI sequencing for the beginner, and it goes much further, dealing with issues that will be of help to more established users. Advice on MIDI fault finding, help for musicians using sequencers with live bands, MIDI messages and MIDI files, plus programming material. So if you are a keen first time user or a MIDI veteran, there will be something in this book to interest you.

CODE B321   **£14.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

MIDI FOR MUSICIANS

by Craig Anderton
How musical instrument digital interface really works, what computers and software are all about, why MIDI is changing the way we record music, and how to make MIDI work for you.

CODE B359   **£10.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

A BEGINNERS GUIDE TO MIDI

by RA Penfold
The title says it all: all aspects of MIDI are explained and many common beginner's problems are discussed.

CODE B287   **£4.95**
Postage: UK £1.50, Europe £3.95, R.O.W. £5.50

COMPUTER & MUSIC

THE DESKTOP MUSCIAN

By David M Rubin
A comprehensive guide to MIDI and audio on PC and Macintosh computers. Purchasers are provided with a CD-ROM full of MIDI software demos to get them going, and the text — over 400 pages of it — covers all aspects of music and computers, from first concepts to advanced arrangement techniques.

CODE B335   **£27.95**
Postage: UK £3.95, Europe £7.50, R.O.W. £14.50

MAKING MUSIC WITH DIGITAL AUDIO

by Ian Waugh
In this practical and clearly-written book, Ian Waugh explains all aspects of the subject from digital audio basics to putting together a system to suit you own music requirements, using the minimum of technical language, the book explains exactly what you need to know. On a practical level you will learn about sample editing, digital manufacturing, digital FX processing, integrating MIDI and digital audio, using sample CDs, mastering to DAT and direct to CD, digital audio and multimedia. 256 pages.

CODE B364   **£14.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

THE MUSICIANS GUIDE TO THE INTERNET

Gary Hustwit
A comprehensive overview of the most popular features of the internet geared specifically for musicians - the complete guide to getting you and your music online!

CODE B362   **£12.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

SYNTHESIZERS

ARRANGING TECHNIQUES FOR SYNTHESISTS

For amateur or professional musicians who wish to increase their knowledge of arranging. Gives you an understanding of acoustic instruments, voices and sound production.

CODE B358   **£10.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

THE A-Z OF ANALOGUE SYNTHS PART ONE: A-M / PART TWO: N-Z

by Peter Forrest
Complete rundown of all the major analogue synths and keyboards ever made, spread across two books. Each is a testament to the massive amount of research and effort put in by the author. Featuring pocket company histories and detailed data on the instruments produced, each book provides a comprehensive overview of the qualities of various instruments; charts and tables assess second hand values and maintenance levels necessary to keep a given instrument playable as well as such intangibles as sound quality, collectability and user interface. Includes colour photos of many classic instruments.

PART ONE: A-M CODE B294 £14.00
Postage: UK £1.95 Europe £5.50 R.O.W. £8.50

PART TWO: N-Z CODE B338 £16.00
Postage: UK £1.95 Europe £5.50 R.O.W. £8.50

VINTAGE SYNTHESIZERS

By Mark Vail
This well-illustrated, 300 page book covers synth history, interviews with designers and overviews of important instruments. Mark Vail's book could be the most entertaining and useful synth book yet – check out the definitive history of the Minimoog, complete with pre-production designers' sketches. While not strictly a buyers' guide, there is a comprehensive section to sourcing, valuing, upgrading and servicing classic instruments.

CODE B199   **ONLY £12.95**
Postage: UK £2.50, Europe £6.95, R.O.W. £9.50

LIVE SOUND

LIVE SOUND REINFORCEMENT

Scott Hunter Stark
A comprehensive guide to P.A. and music reinforcement systems and technology. From basic principles to nuances in the art of engineering, this is the definitive work on live sound and all its users.

CODE B361   **£19.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

LIVE SOUND

Peter Buick has produced a book written for live sound engineers and gigging musicians with practical applications and real life scenarios very much in mind. The whole ambit of live sound engineering is addressed, including the equipment used and how to get the most out of it, the problems you will confront and how to fix them, and various other related subjects such as safety, feedback, as well as automation and MIDI in the live sound field.

CODE B337   **£14.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

SOUND CHECK

Sound Check is a simplified guide to what can be a tricky subject: an easy-to-understand explanation of the principles and physics of sound. Sound Check goes on to cover amplifiers, speaker hookup, matching speakers with amps, sound reinforcement, mixers, monitor systems, grounding and more. Includes many diagrams, lots of illustrations, and a glossary of terms for audio made easy!

CODE B357   **£8.95**
Postage: UK £2.50, Europe £5.50, R.O.W. £8.50

World Radio History

World Radio History

World Radio History

MUSIC CONTROL HQ
CHAPEL MEWS,
CREWE RD, ALSAGER,
STAFFS. ST7 2HA

T: 01270 863 779
F: 01270 8E3 847

MUSIC CONTROL SOUTH
43-44 HOXTON SQUARE,
LONDON. N16 PB

T: 0171 729 4100
F: 0171 729 7400

MUSIC CONTROL SURREY

T: 01252 715700

EMAIL:
chris/emma/daz
@music-control.co.uk
www.music-control.co.uk

BRAND NEW PRODUCTS

AKAI S5000	£ 1799.00
AKAI S6000	£ 2799.00
ALESIS PCR PC/MAC	£ 499.00
LOGIC AUDIO PLAT.	£ 549.00
LOGIC AUDIO GOLD	£ 399.00
LOGIC AUDIO SILVER	£ 199.00
EMU AUDITY 2000	£ 1299.00
FOSTEX MD4	£ 399.00
ROLAND SP808	£ 1099.00
ROLAND JX305	£ 949.00
ROLAND VS1680	£ 2199.00
SPIRIT DIGITAL 328	£ 3499.00
YAMAHA EX5	£ 1899.00
YAMAHA EX5R	£ 1599.00

NEW EQUIPMENT

AKAI S2000	£ 649.00
AKAI S3000 XL	£ 1099.00
AKAI MPC2000	£ 899.00
APPLE MAC G3 SERIES	£ CALL
NORD LEAD II RACK	£ 1295.00
NORD LEAD MODULAR	£ CALL
DRAWMER 1962	£ 1299.00
DRAWMER MX40	£ 359.00
DRAWMER MX50	£ 269.00
DIGITECH ST 4 QUAD	£ CALL
EMAGIC AUDIOWERK 8	£ 369.00
EMU E-SYNTH	£ CALL
EMU E4XTURBO 2gb	£ CALL
EMU ESI4000 Turbo	£ CALL
EMU E6400	£ LOW
EVENTIDE H3000DSE	£ 1849.00
FOSTEX D90	£ 1499.00
FOSTEX DMT8VL V2	£ CALL
KORG TRINITY PLUS	£ CALL
KORG TR RACK	£ 999.00
KORG 1212 I/O	£ 539.00
LEXICON PCM81	£ CALL
LEXICON PCM91	£ CALL
LEXICON MPX100	£ 249
MACKIE 8 8US	£ CALL
PANASONIC SV3800	£ CALL
ROLAND JV2080	£ 1039.00
ROLAND JV1080	£ 749.00
ROLAND VS880CO	£ CALL
ROLAND VS840	£ 949.00
SPIRIT FOLIO SX	£ 399.00
YAMAHA 01V	£ 1399.00

USED EQUIPMENT

		HOHNER CLAVINET D6	£ 449.00	ROLAND JX8P	£ 299.00
		KAWAI K1	£ 199.00	ROLAND JX3P	£ CALL
AKAI S01	£ CALL	KAWAI K4R	£ 279.00	ROLAND MC202	£ 279.00
AKAI S900	£ 299.00	KORG 01W FD	£ 699.00	ROLAND MKS7	£ 279.00
AKAI S950	£ 399.00	KORG BX3	£ CALL	ROLAND MKS50	£ 379.00
AKAI S1000	£ 799.00	KORG CX3	£ 679.00	ROLAND MKS80	£ 899.00
AKAI S3000	£ CALL	KORG DW8000	£ 299.00	ROLAND PG380	£ 249.00
AKAI CD3000i	£ 749.00	KORG M1	£ 499.00	ROLAND PG800	£ CALL
ALESIS QUAD/VERB +	£ 249.00	KORG M3R	£ 299.00	ROLAND R8	£ 379.00
ALESIS MIDIVERB IV	£ 299.00	KORG MONOPOLY	£ 449.00	ROLAND 201/301/501	£ CALL
ALESIS MMT8	£ 149.00	KORG MS10	£ 249.00	ROLAND S8E555	£ CALL
ARP 2500	£ CALL	KORG MS20	£ CALL	ROLAND SVC350	£ 549.00
ARP 2600	£ CALL	KORG POLY 6	£ 279.00	ROLAND SYSTEM 100	£ 1499.00
ARP MODULAR	£ CALL	KORG TRINITY	£ CALL	ROLAND S750/760/770	£ CALL
ARP ODYSSEY	£ 549.00	KORG VC10	£ 449.00	ROLAND SH2	£ CALL
ARP 2600	£ CALL	KORG WAVESTATION	£ 529.00	ROLAND SH101	£ 279.00
ARP OMNI 1	£ CALL	KORG W/STATION SR	£ 449.00	ROLAND SE70	£ CALL
ARP OMNI II	£ 199.00	KORG X3	£ CALL	ROLAND SE50	£ CALL
ARP SEQUENCER	£ 499.00	KURZWEIL K2000	£ CALL	ROLAND TB303	£ CALL
ARP SOLINA	£ CALL	LEXICON PCM70	£ CALL	ROLAND TR808	£ CALL
CASIO FZ1	£ 429.00	MACKIE CR1604	£ 549.00	ROLAND TR606	£ 149.00
CASIO FZ10M	£ 399.00	MOOG PRODIGY	£ 299.00	ROLAND TR909	£ CALL
OBUMAR SPIRIT	£ 379.00	MOOG MODULAR	£ CALL	RHODES CHROMA	£ 849.00
CHEETAH MS6	£ 199.00	MOOG MINI MOOG	£ CALL	SEQUENTIAL DRUMTRX	£ CALL
DRAWMER DS201	£ 229.00	OBERHEIM M1000	£ 329.00	SEQUENTIAL PRO 1	£ 429.00
DRAWMER DL221	£ 229.00	OBERHEIM M6/6R	£ CALL	SEQUENTIAL PRO V	£ 1099.00
DRAWMER DF320	£ 349.00	OBERHEIM OBXA	£ CALL	SEQUENTIAL PRO 600	£ 599.00
DRAWMER DL251	£ 449.00	OBERHEIM 4 VOICE	£ CALL	SEQUENTIAL SIXTRAK	£ 379.00
DRAWMER LX20	£ 149.00	OBERHEIM OB8	£ 949.00	TASCAM DA30	£ CALL
EDP WASP	£ 349.00	OCTAVE CAT	£ 420.00	TASCAM DA88	£ 1849.00
ELKA SYNTHEX	£ CALL	OSC OSCAR	£ 799.00	TEISCO 110F	£ 299.00
EMU ESI 32	£ 599.00	PEAVEY SPEC BASS	£ 99.00	YAMAHA CS5	£ 189.00
EMU E64	£ CALL	POLYFUSION MODULAR	£ CALL	YAMAHA CS15	£ 349.00
EMU MORPHEUS	£ 499.00	RHODES CHROMA	£ 899.00	YAMAHA CS40M	£ CALL
EMU PROTEUS 1	£ 249.00	ROLAND CR78	£ 249.00	YAMAHA DX100	£ 199.00
EMU PROTEUS 2	£ 379.00	ROLAND D50	£ CALL	YAMAHA DX7	£ CALL
EMU SOUND ENGINE	£ 99.00	ROLAND JD800	£ 799.00	YAMAHA SPX1000	£ CALL
EMU SP1200	£ CALL!	ROLAND JP6	£ 749.00	YAMAHA SPX90	£ 229.00
ENSONIQ DP2	£ CALL	ROLAND JP8	£ 999.00	YAMAHA SPX90 II	£ 299.00
ENSONIQ DP4 +	£ CALL	ROLAND D70	£ CALL	YAMAHA SY77	£ CALL
FARFISA COMPACTDUO	£ CALL	ROLAND JUNO 2	£ 349.00	YAMAHA SY85	£ 629.00
FENDER RHODES 73	£ CALL	ROLAND JUNO 106	£ CALL	YAMAHA TX81Z	£ 199.00
FOSTEX R42	£ 399.00	ROLAND JUNO 60	£ 349.00	YAMAHA TX816	£ CALL
FOSTEX R8	£ 549.00	ROLAND JX10	£ 549.00	YAMAHA TX802	£ CALL

music control

01270 883779

World Radio History

World Radio History

World Radio History

SITUATIONS VACANT

RECORDING STUDIOS

CLUBS

FLIGHT CASES

World Radio History

World Radio History

sounding Off

'That MARTIN WALKER', you probably think as you leaf through your new *SOS*, 'he must be a happy man. He's always getting to look at the very latest music software — and for free!' But no, he's complaining about it. Here's why...

hy is it that some music software is so difficult to use? After battling for ages to achieve what you set out to do, you often have the feeling that you've managed it despite the software, rather than because of it. Occasionally, I've even had to resort to writing down step-by-step details of how I finally managed to get something convoluted to work, because having to go through it all again doesn't bear thinking about.

Sometimes, it seems that a program is determined not to let you find an easy way to do what seems to you basic and obvious. The solution may still be only a keyclick away, but unless the manual and on-line help can point you in the right direction, you have to rely on the intuitiveness of the user interface. Whatever the complexity of the underlying software engine, it is this interface that makes or breaks software as far as the user is concerned.

Of course, this is just as important when designing hardware, but here there are other constraints. On a typical synth module, for instance, there is a limit to how many knobs and buttons will fit on the designated size of front panel. However, the software user interface need suffer from no such limitations; virtual front panels can be programmed in many different ways. So why is some software still so difficult to use?

Perhaps the answer lies in the way software is put together today. Many modern applications are so huge that they are under continual development by teams of programmers, with several different groups working on different aspects of the whole package. This can make it difficult to keep an overall view; even if one group has a great idea to make their part of the program easier to use, it may be discarded for the sake of keeping a consistent user interface.

Another, more fundamental aspect is that the programmers themselves are often too close to their work — too focused on algorithms and bug-hunting — to see a more elegant way to the same end. The task of altering software with an over-complicated user interface to one that is refreshingly simple is also a pretty thankless one. An elegant, easy-to-use interface hides a huge amount of extra work — the simplicity of the end result is largely because of all the extra hidden software intelligence doing additional work under the surface. Such transparent simplicity can even work against a product, since a huge number of obvious new features is likely to sell more units than newly-found simplicity and ease of use.

And so we end up with situations where some developers are so proud of their software engines that they scorn anyone who is not prepared to battle though the impregnable user interface. Of course, these tend to be the applications that end up with large libraries of presets — users want the end results, but few have the time or the inclination to devote large number of hours locked away with the manual to work things out for themselves.

I can see the problem from both sides, as I have been a software and hardware developer, but am also in the privileged position of seeing much of the latest PC software for review in *SOS*. It saddens me to see software that contains so many clever ideas lost behind an unwieldy user interface, and conversely, software selling on the basis of its amazing graphics, even though there can be little of substance beneath.

Revolutionary interfaces, such as the original Steinberg *Cubase* Arrange page, tend to be designed from scratch by extremely talented individuals or a small group of programmers with a vision. However, once software gets large enough to acquire an inertia of its own, it is almost impossible to make such revolutionary changes any more — there are simply too many other features that would have to be altered in order to work with the new concepts. So we are left with evolution rather than revolution, and people become locked into using a particular piece of software simply because despite its obvious faults and limitations, the thought of starting again with a new package just doesn't bear thinking about.

Next time you are tempted to buy a new piece of software, don't just look at the list of features. Long after their gloss has worn off, it will be the user interface that counts — that thin veneer between man and machine that sometimes wears a bit too thin. **SOS**